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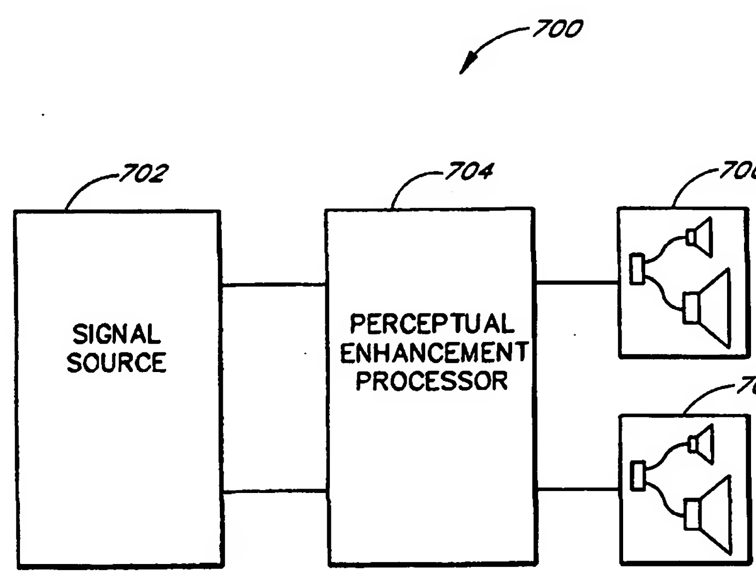


**(57) Abstract**

The present invention provides an audio enhancement apparatus and method which spectrally shapes harmonics of the low-frequency information in a pair of audio signals so that when reproduced by a loudspeaker, a listener perceives the loudspeaker as having more acoustic bandwidth than is actually provided by the loudspeaker. The perception of extra bandwidth is particularly pronounced at low frequencies, especially frequencies at which the loudspeaker system produces less acoustic output energy. In one embodiment, the invention also shifts signal from one audio signal to the other audio signal in order to obtain more bandwidth for the available loudspeaker to reduce clipping. In one embodiment, the invention also provides a combined signal path for spectral shaping of the desired harmonics and a feedforward signal path for each pair of audio signals.

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# LOW-FREQUENCY AUDIO SIMULATION SYSTEM

## Field of the Invention

This invention relates generally to audio enhancement systems and methods for improving the realism of sound reproduction. More particularly, this invention relates to apparatus and methods for enhancing the perceived low-frequency content of acoustic energy produced by an acoustic transducer, such as a loudspeaker.

## Background

The audio and multimedia industries have continually struggled to overcome the imperfections of reproduced sound. For example, it is often difficult to adequately reproduce low-frequency sounds such as bass. Various conventional approaches to improving the output of low-frequency sounds include the use of higher quality speakers with greater cone areas, larger magnets, larger housings, or greater cone excursion capabilities. In addition, conventional systems have attempted to reproduce low-frequency sounds with resonant chambers and horns which match the acoustic impedance of the loudspeaker to the acoustic impedance of free space surrounding the loudspeaker.

Not all systems, however, can simply use more expensive or more powerful speakers to reproduce low-frequency sounds. For example, some conventional sound systems such as compact audio systems and multimedia computer systems rely on small loudspeakers. In addition, to conserve costs, many audio systems use less accurate loudspeakers. Such loudspeakers typically do not have the capability to properly reproduce low-frequency sounds and consequently, the sounds are typically not as robust or enjoyable as systems which more accurately reproduce low-frequency sounds.

Some conventional enhancement systems attempt to compensate for poor reproduction of low-frequency sounds by amplifying the low-frequency signals prior to inputting the signals into the loudspeakers. Amplifying the low-frequency signals delivers a greater amount of energy to the loudspeakers which in turn, drives the loudspeakers with greater forces. Such attempts to amplify the low-frequency signals, however, can result in overdriving the loudspeakers. Unfortunately, overdriving the loudspeakers can increase the background noise, introduce distracting distortions, and damage the loudspeakers.

Still other conventional systems, in an attempt to compensate for the lack of the lower-frequencies, distort the reproduction of the higher frequencies in ways that add undesirable sound coloration.

## Summary of the Invention

The present invention provides a unique apparatus and method which enhances the perception of low-frequency sounds. In loudspeakers which do not reproduce certain low-frequency sounds, the invention creates the illusion that the missing low-frequency sounds do exist. Thus, a listener perceives low-frequencies which are below the frequencies the loudspeaker can actually reproduce. This illusionary effect is accomplished by exploiting, in a unique manner, how the human auditory system processes sound.

One embodiment of the invention exploits how a listener mentally perceives music or other sounds. The process of sound reproduction does not stop at the acoustic energy produced by the loudspeaker, but includes the ears, auditory nerves, brain, and thought processes of the listener. Hearing begins with the action of the ear and the auditory nerve system. The human ear may be regarded as a delicate translating system which receives acoustical vibrations, converts these vibrations into nerve impulses, and ultimately into the "sensation" or perception of sound.

The human ear is known to be non-linear in its response to acoustic energy. This non-linearity of the hearing mechanism produces intermodulation distortion in the form of additional overtones and harmonics which do not exist in the actual program material. These non-linear effects are particularly pronounced at low frequencies and these effects have a pronounced effect on how low-frequency sounds are perceived.

Advantageously, some embodiments of the invention optimally exploit how the human ear processes overtones and harmonics of low-frequency sounds to create the perception that non-existent low-frequency sounds are being emitted from a loudspeaker. In some embodiments the frequencies in higher-frequency signals are selectively emphasized to create the illusion of lower-frequency signals. In other embodiments, certain higher-frequency bands are modified with a plurality of filter functions.

In addition, some embodiments of the invention are designed to improve the low-frequency enhancement of popular audio program material, such as music. Most music is rich in harmonics. Accordingly, these embodiments can modify a wide variety of music types to exploit how the human ear processes low-frequency sounds. Advantageously, music in existing formats can be simply processed to produce the desired effects.

This entirely new approach produces a number of significant advantages. Because a listener perceives low-frequency sounds which do not actually exist, the need for large speakers, greater cone excursions, or added horns is reduced. Thus, in one embodiment, small loudspeakers can appear as if they are emitting the low-frequency sounds of larger speakers. As can be expected, this embodiment produces the perception of low-frequency audio such as bass, in sound environments which are too small for large loudspeakers.

In addition, with one embodiment of the invention, the small loudspeakers in hand-held and portable sound systems can create a more enjoyable perception of low-frequency sounds. Thus one need not sacrifice low-frequency sound quality for portability.

In one embodiment of the invention, lower-cost speakers create the illusion of low-frequency sounds. Many low-cost loudspeakers cannot adequately reproduce low-frequency sounds. Rather than actually reproducing low-frequency sounds with expensive speaker housings, high performance components and large magnets, one embodiment uses higher frequency sounds to create the illusion of low-frequency sounds. As a result, lower-cost speakers can be used to create a more realistic and robust listening experience.

Furthermore, in one embodiment, the illusion of low-frequency sounds creates a heightened listening experience which increases the realism of the sound. Thus, instead of the reproduction of the muddy or wobbly

low-frequency sounds existing in many low-cost prior art systems, one embodiment of the invention reproduces sounds which are perceived to be more accurate and clear. Such low-cost audio and audio-visual devices can include, by way of example, radios, mobile audio systems, computer games, compact disk (CD) players, digital versatile disk (DVD) players, multimedia presentation devices, computer sound cards, and the like.

In one embodiment, creating the illusion of low-frequency sounds requires less energy than actually reproducing the low-frequency sounds. Thus, systems which operate on batteries or in low-power environments, can create the illusion of low-frequency sounds without consuming as much valuable energy as systems which simply amplify or boost low-frequency sounds.

Other embodiments of the invention create the illusion of lower-frequency signals with specialized circuitry. These circuits are simpler than prior art low-frequency amplifiers and thus reduce the costs of manufacturing. Advantageously, these cost less than prior art sound enhancement devices which add complex circuitry.

Still other embodiments of the invention rely on a microprocessor which implements the disclosed low-frequency enhancement techniques. In some cases, existing processing audio components can be reprogrammed to provide the disclosed unique low-frequency signal enhancement techniques of one or more embodiments of the invention. As a result, the costs of adding low-frequency enhancement to existing systems is significantly reduced.

In one embodiment, the sound enhancement apparatus receives at least two input signals, from a host system and in turn, generates two enhanced output signals. In particular, the two input signals are processed to provide a pair of spectrally enhanced output signals, that when played on a loudspeaker and heard by a listener, produce the sensation of extended bass. In addition, the embodiment modifies the low-frequency audio information in a different manner than the high-frequency audio information.

In one embodiment, the sound enhancement apparatus receives one or more input signals and generates one or more enhanced output signals. In particular, the input signals comprise waveforms having a first frequency range and a second frequency range. The input signals are processed to provide the enhanced output signals, that when played on a loudspeaker and heard by a listener, produce the sensation of extended bass. In addition, the embodiment may modify information in the first frequency range in a different manner than information in the second frequency range. In some embodiments, the first frequency range may be bass frequencies too low for the desired loudspeaker to reproduce and the second frequency range may be midbass frequencies that the loudspeaker can reproduce.

One embodiment modifies the audio information which is common to two stereo channels in a manner different from energy which is not common to the two channels. The audio information which is common to both input signals is referred to as the combined signal. In one embodiment, the enhancement system spectrally shapes the amplitude of the phase and frequencies in the combined signal in order to reduce the clipping which may result from high-amplitude input signals without removing the perception that the audio information is in stereo.

As discussed in more detail below, one embodiment of the sound enhancement system spectrally shapes the combined signal with a variety of filters to create an enhanced signal. By enhancing selected frequency bands within the combined signal, the embodiment provides a perceived loudspeaker bandwidth which is wider than the actual loudspeaker bandwidth.

One embodiment of the sound enhancement apparatus includes feedforward signal paths for the two stereo channels and four parallel filters for the combined signal path. Each of the four parallel filters comprises a sixth order bandpass filter consisting of three series connected biquad filters. The transfer functions for these four filters are specially selected to provide phase and/or amplitude shaping of various harmonics of the low-frequency content of an audio signal. The shaping unexpectedly increases the perceived bandwidth of the audio signal when played through loudspeakers. In another embodiment, the sixth order filters are replaced by lower order Chebychev filters.

Because the spectral shaping occurs on the combined signal, which is then combined with the stereo information in the feedforward paths, the frequencies in the combined signal can be altered such that both stereo channels are affected, and some signals in certain frequency ranges are coupled from one stereo channel to the other stereo channel. As a result, the preferred embodiment can create enhanced audio sound in an entirely unique, novel, and unexpected manner.

The sound enhancement apparatus may in turn, be connected to one or more subsequent signal processing stages. These subsequent stage may provide improved soundstage or spatial processing. The output signals can also be directed to other audio devices such as recording devices, a power amplifiers, loudspeakers, and the like without affecting the operation of the sound enhancement apparatus.

In yet another embodiment, the sound enhancement is provided by a signal processor configured to generate a second set of frequencies from an input signal that has a a first set of frequencies. The signal processor may be implemented as hardware, software (e.g., in a DSP), or both. The second set of frequencies is generated so as to create the perception that the second set of frequencies contains at least some of the harmonics of the first set of frequencies. The signal processor uses a zero crossing detector driving a monostable multivibrator to provide a series of pulses. The pulses are created by zero crossings of the input signal corresponding to the first set of frequencies. The signal processor generates the second set of frequencies by delivering the series of pulses to a collection of bandpass filters.

In yet another embodiment, the sound enhancement is provided by a signal processor configured to process the input signal through a collection of bandpass filters. An output of selected bandpass filters is provided to an input of an automatic gain control (AGC) amplifier, such that each selected bandpass filter drives a separate AGC amplifier. Each AGC amplifier has a control input that sets the output level of the amplifier. The control inputs are set in response to the low frequency content of the input signal. The signal processor may be

implemented as hardware, software (e.g., in a DSP), or both. The outputs of the AGC amplifiers are added together to produce an enhanced output signal.

In some embodiments, input signals are combined to produce a combined signal which is then enhanced to produce an enhanced combined signal. The enhanced combined signal is then combined with each of the original input signals to produce the output signals. In other embodiments, the input signals are not combined, but kept separate. The separate input signals are each enhanced separately to produce enhanced output signals. The same signal processing may be used to enhance the combined signal or the separate input signals.

## Brief Description of the Drawings

These and other aspects, advantages, and novel features of the invention will become apparent upon reading the following detailed description and upon reference to the accompanying drawings.

Figure 1 is a block diagram of an audio system appropriate for use with the present invention.

Figure 2 is a block diagram of a multimedia computer system having a sound card and loudspeakers.

Figure 3 is a plot of the frequency response of a typical small loudspeaker system.

Figure 4A illustrates the actual and perceived spectrum of a signal represented by two discrete frequencies.

Figure 4B illustrates the actual and perceived spectrum of a signal represented by a continuous spectrum of frequencies.

Figure 4C illustrates a time waveform of a modulated carrier.

Figure 4D illustrates the time waveform of Figure 4C after detection by a detector.

Figure 6A is a block diagram of a digital sound system.

Figure 6B is a block diagram of a digital sound system with sound enhancement processing.

Figure 7 is a block diagram of a sound card with external sound enhancement processing.

Figure 8 illustrates one embodiment of the signal processing used to shape the spectrum of an input signal to enhance the perception of low-frequency sounds.

Figure 9 is a circuit diagram of a bandpass filter used in some embodiments of the present invention.

Figure 10 is a plot of the transfer functions of the bandpass filters used in the signal processing diagram shown in Figure 8.

Figure 11 is a signal processing block diagram of a perceptual enhancement system that uses a zero crossing detector.

Figure 12A illustrates an enhancement transfer function which has been generated using a number of automatic gain control circuits connected to the bandpass filters shown in Figure 8, the enhancement transfer function corresponding to an input signal having significant low-frequency energy.

Figure 12B illustrates the resulting total spectrum produced by the enhancement transfer function shown in Figure 12A.

Figure 12C illustrates an enhancement transfer function which has been generated using a number of automatic gain control circuits connected to the bandpass filters shown in Figure 8, the enhancement transfer function corresponding to an input signal with very little low-frequency energy.

Figure 12D illustrates the resulting total spectrum produced by the enhancement transfer function shown in Figure 12C.

Figure 13 is a signal processing block diagram of a system that produces the enhancement transfer functions shown in Figure 12.

Figure 14A is a block diagram of an automatic gain control amplifier.

Figure 14B is a circuit diagram of an automatic grain control amplifier corresponding to the block diagram shown in Figure 14A.

Figure 15 is a signal processing block diagram of a system that provides enhancement transfer functions as shown in Figure 12 with selectable frequency response.

## Detailed Description of the Preferred Embodiment

The present invention provides a method and system for enhancing audio signals. The sound enhancement system improves the realism of sound with a unique sound enhancement process. Generally speaking, the sound enhancement process receives two input signals, a left input signal and a right input signal, and in turn, generates two enhanced output signals, a left output signal and a right output signal.

The left and right input signals are processed collectively to provide a pair of left and right output signals. In particular, the enhanced system embodiment equalizes the differences which exist between the two input signals in a manner which broadens and enhances the perceived bandwidth of the sounds. In addition, many embodiments adjust the level of the sound which is common to both input signals so as to reduce clipping. Advantageously, some embodiments achieve sound enhancement with simplified, low cost, and easy-to-manufacture analog circuits that do not require digital signal processing.

Although the embodiments are described herein with reference to a preferred sound enhancement system, the invention is not so limited, and can be used in a variety of other contexts in which it is desirable to adapt different embodiments of the sound enhancement system to different situations.

### Overview Of A Sound Enhancement System

Figure 1 is a block diagram of a sound enhancement system 100 comprising a sound enhancement apparatus 104. The sound enhancement system 100 includes a sound source 102, the sound enhancement apparatus 104, an optional signal processing system 106, an optional amplifier 108, loudspeakers 110, and a listener 112. Typically, the sound source 102 is an audio signal source that generates a stereo signal comprising a left channel and a right channel.

The signal source 102 can include, by way of example, a stereo receiver, radio, compact disc player, video cassette recorder (VCR), audio amplifiers, theater systems, televisions, laser disc players, digital versatile disk (DVD) players, devices for recording and playback of prerecorded audio, multimedia devices, computer games and the like. While the signal source 102 typically generates a set of stereo signals, it should be understood that the signal source 102 is not limited to stereo signals. Thus, in other embodiments, the signal source 102 can generate a wide variety of audio signals such as audio systems which generate multi-channel signals.

The signal source 102 transmits the left and right channels to the sound enhancement system 104. The sound enhancement system 104 enhances the low-frequency audio information through modification of the left and right channels. In other embodiments, the left and right channel input signals need not be stereo signals and can include a wide range of audio signals, such as a Dolby Laboratories Pro-Logic system which uses a matrixing scheme to store four or more separate audio channels on just two audio recording tracks. The audio signals can also include surround sound systems which can deliver completely separate forward and rear audio channels. One such system is Dolby Laboratories five-channel digital system dubbed "AC-3."

The audio information, which is the sum of the left and right channels, is referred to as the combined information, or the combined signal. One embodiment shapes the spectral harmonics of the frequencies in the combined signal, and then inserts portions of the shaped combined signal back into the left and right channels in order to reduce the clipping which may result from low-frequency, high-amplitude input signals in one channel or the other.

Figure 2 illustrates a typical multimedia computer system 200 which may advantageously use an embodiment of the present invention to improve the audio performance produced by a pair of small desktop computer loudspeakers 210. The loudspeakers 210 are connected to a plug-in card 206 inside a computer unit 204. The plug-in card 206 will typically be a sound card such as the sound card shown in Figure 5, but may also be any computer interface card that produces audio output, including a radio card, television tuner card, internal modem, plug-in Digital Signal Processor (DSP) card, etc. A computer user 202 uses the computer 204 to run a computer program which causes the plug-in card 206 to generate audio signals that are converted by the loudspeakers 210 into acoustic waves.

The loudspeakers 210 used by a multimedia computer system are typically small desktop units that are designed to be small and inexpensive, and therefore do not have the capability to produce significant sound pressure levels at low frequencies. A typical small loudspeaker system used for multimedia computers will have an acoustic output response that rolls off at about 200 Hz. Figure 3 shows the measured response 308 of a typical small computer loudspeaker system that uses a high-frequency driver (tweeter) to reproduce the high frequencies, and a four inch midrange-bass driver (woofer) to reproduce the midrange and bass frequencies. Such a system employing two drivers is often called a two-way system. Loudspeaker systems employing more than two drivers are known in the art and will work with an embodiment of the present invention. Loudspeaker systems with a

single driver are also known and will also work with the present invention. The response 308 is plotted on a rectangular plot with an X-axis showing frequencies from 20 Hz to 20 kHz. This frequency band corresponds to the range of normal human hearing. The Y-axis in Figure 3 shows normalized amplitude response from 0 dB to -50 dB. The curve 308 is relatively flat in a midrange frequency band from approximately 2 kHz to 10 kHz, showing some rolloff above 10 kHz. In the low frequency ranges, the curve 308 exhibits a low-frequency rolloff that begins in a midbass band between approximately 200 Hz and 2 kHz such that below 200 Hz, the loudspeaker system produces very little acoustic output.

Many cone-type drivers are very inefficient when producing acoustic energy at low frequencies where the diameter of the cone is less than the wavelength of the acoustic sound wave. When the cone diameter is smaller than the wavelength, maintaining a uniform sound pressure level of acoustic output from the cone requires that the cone excursion be increased by a factor of four for each octave (factor of 2) that the frequency drops. The maximum allowable cone excursion of the driver is quickly reached if one attempts to improve low-frequency response by simply boosting the electrical power supplied to the driver.

Thus, the low-frequency output of a driver cannot be increased beyond a certain limit, and this explains the poor low-frequency sound quality of most small loudspeaker systems. The curve 308 is typical of most small loudspeaker systems that employ a low-frequency driver of approximately four inches in diameter. Loudspeaker systems with larger drivers will tend to produce appreciable acoustic output down to frequencies somewhat lower than those shown in Figure 3, and systems with smaller low-frequency drivers will typically not produce output as low as that shown in Figure 3.

As discussed above, to date, a system designer has had little choice when designing loudspeaker systems with extended low-frequency response. Previously known solutions were expensive and produced loudspeakers that were too large for the desktop. One popular solution to the low-frequency problem is the use of a sub-woofer, which is usually placed on the floor near the computer system. Sub-woofers can provide adequate low-frequency output, but they are expensive, and thus relatively uncommon as compared to inexpensive desktop loudspeakers.

Rather than use drivers with large diameter cones, or a sub-woofer, an embodiment of the present invention overcomes the low-frequency limitations of small systems by using characteristics of the human hearing system to produce the perception of low-frequency acoustic energy, even when such energy is not produced by the loudspeaker system.

The human auditory system is known to be non-linear. A non-linear system is, simply put, a system where an increase in the input is not followed by a proportional increase in the output. Thus, for example, in the ear, a doubling of the acoustic sound pressure level does not produce a perception that the volume of the sound source has been doubled. In fact, the human ear is, to a first approximation, a square-law device that is responsive to power rather than intensity of the acoustic energy. This non-linearity of the hearing mechanism produces

intermodulation frequencies that are heard as overtones or harmonics of the actual frequencies in the acoustic wave.

The intermodulation effect of the non-linearities in the human ear is shown in Figure 4A, which illustrates an idealized amplitude spectrum of two pure tones. The spectral diagram in Figure 4A shows a first spectral line 404 which corresponds to acoustic energy produced by a loudspeaker driver (e.g., a sub-woofer) at 50 Hz. A second spectral line 402 is shown at 60 Hz. The lines 404 and 402 are actual spectral lines corresponding to real acoustic energy produced by the driver, and no other acoustic energy is assumed to exist. Nevertheless, the human ear, because of its inherent non-linearities, will produce intermodulation products corresponding to the sum of the two actual spectral frequencies and the difference between the two spectral frequencies.

For example, a person listening to the acoustic energy represented by the spectral lines 404 and 402 will perceive acoustic energy at 50 Hz, as shown by the spectral line 406, at 60 Hz, as shown by the spectral line 406, and at 110 Hz, as shown by the spectra line 410. The spectral line 410 does not correspond to real acoustic energy produced by the loudspeaker, but rather corresponds to a spectral line created inside the ear by the non-linearities of the ear. The line 410 occurs at a frequency of 110 Hz which is the sum of the two actual spectral lines (110 Hz = 50 Hz + 60 Hz). Note that the non-linearities of the ear will also create a spectral line at the difference frequency of 10 Hz ( 10 Hz = 60 Hz - 50 Hz), but that line is not perceived because it is below the range of human hearing.

Figure 4A illustrates the process of intermodulation inside the human ear, but it is somewhat simplified when compared to real program material, such as music. Typical program material such as music is rich in harmonics, so much so that most music exhibits an almost continuous spectrum, as shown in Figure 4B. Figure 4B shows the same type of comparison between actual and perceived acoustic energy, as shown in Figure4A, except that the curves in Figure 4B are shown for continuous spectra. Figure 4B shows an actual acoustic energy curve 420 and the corresponding perceived spectrum 430.

As with most non-linear systems, the non-linearity of the ear is more pronounced when the system is making large excursions (e.g., large signal levels) than for small excursions. Thus, for the human ear, the non-linearities are more pronounced at low frequencies, where the eardrum and other elements of the ear make relatively large mechanical excursions, even at lower volume levels. Thus, the figure 4B shows that the difference between actual acoustic energy 420, and the perceived acoustic energy 430 tends to be greatest in the low-frequency range and becomes relatively smaller at the higher frequencies.

As shown in Figures 4A and 4B, low-frequency acoustic energy comprising multiple tones or frequencies will produce, in the listener, the perception that the acoustic energy in the midbass range contains more spectral content than actually exists. The human brain, when faced with a situation where information is thought to be missing, will attempt to "fill in" missing information on a subconscious level. This filling in phenomenon is the basis for many optical illusions. In an embodiment of the present invention, the brain can be tricked into filling in low-

frequency information that is not really present by providing the brain with the midbass effects of such low-frequency information.

In other words, if the brain is presented with the harmonics that would be produced by the ear if the low-frequency acoustic energy was present (e.g., the spectral line 410) then under the right conditions, the brain will subconsciously fill in the low-frequency spectral lines 406 and 408 which it thinks "must" be present. This filling in process is augmented by another effect of the non-linearity of the human ear known as the detector effect.

The non-linearity of the human ear also causes the ear to act like a detector, similar to a diode detector in an Amplitude Modulation (AM) receiver. If a midbass harmonic tone is AM modulated by a low-frequency tone, the ear will demodulate the modulated midbass carrier to produce the low-frequency envelope. Figures 4C and 4D graphically illustrate the modulated and demodulated signal. Figure 4C shows, on a time axis, a modulated signal comprising a higher-frequency carrier signal (e.g. the midbass carrier) modulated by a low-frequency signal.

The amplitude of the higher-frequency signal is modulated by a lower frequency tone, and thus, the amplitude of the higher-frequency signal varies according to the frequency of the lower frequency tone. The non-linearity of the ear will partially demodulate the signal such that the ear will detect the low-frequency envelope of the higher-frequency signal, and thus produce the perception of the low-frequency tone, even though no actual acoustic energy was produced at the lower frequency. As with the intermodulation effect discussed above, the detector effect can be enhanced by proper signal processing of the signals in the midbass frequency range, typically between 100-200 Hz on the low end of the range and 500 Hz on the high end of the range. By using the proper signal processing, it is possible to design a sound enhancement system that produces the perception of low-frequency acoustic energy, even when using loudspeakers that are incapable of producing such energy.

The perception of the actual frequencies present in the acoustic energy produced by the loudspeaker may be deemed a first order effect. The perception of additional harmonics not present in the actual acoustic frequencies, whether such harmonics are produced by intermodulation distortion or detection, may be deemed a second order effect.

Before describing the details of the actual signal processing used in a sound enhancement system, it is helpful to examine several implementations of the system. The sound enhancement system is by no means limited to multimedia computer systems and may be used with many sources of audio signals and many different types of loudspeakers. However, the popularity of multimedia computer systems with inadequate loudspeakers, and the possibility of implementing the sound enhancement system as a software upgrade to the multimedia computer, makes the multimedia computer an attractive platform for several embodiments of the present invention.

Figure 5 is a block diagram illustrating a typical multimedia computer system 500 having a sound card 510, a first loudspeaker system 512, and a second loudspeaker system 514. The computer system 500 comprises a data storage medium 506, a processor 502, and the sound card 510, all connected to an input/output (I/O) bus 508. A main memory 504 for storing programs and data is typically connected to the processor 502 by a separate

memory bus. The sound card 510 comprises an I/O control module 520 which is connected to the data bus 508 and provides the necessary functionality to communicate with the data bus 508. Within the sound card 510, a bi-directional data path connects the I/O control module 520 to a data router 522, which provides multiplexing and demultiplexing of data from the various internal data paths of the sound card and the I/O control module 520.

A first output of the router 522 provides data to a first synthesis module 524 which generates sounds, usually by either FM synthesis or wavetable synthesis. An output of the first synthesis module 524 is fed through a first gain control 534 to a first mixer (adder) 528. A second output of the router 522 provides data to an input of a first Digital Signal Processor (DSP) 525. An output of the first DSP 525 is provided to an input of a first digital-to-analog converter (DAC) 526. The DSP 525 is optional and not found on all sound cards. On cards without the DSP 525, an output of the router 522 may be connected directly to the input of the first digital-to-analog converter 526. An output of the first DAC 526 is connected through a gain control 536 to an input of the mixer 528. An output of the mixer 528 is connected through a gain control 530 to a first power amplifier 520. An output of the first power amplifier 520 is provided to the loudspeaker system 512.

A third output of the router 522 provides data to a second synthesis module 544. An output of the second synthesis module 544 is fed through a gain control 554 to a second mixer 548. A third output of the router 522 provides data to an input of a second Digital Signal Processor (DSP) 545. An output of the second DSP 545 is provided to an input of a second DAC 526. The DSP 545 is optional, and if not provided, an output of the router 522 may be connected directly to the input of the second DAC converter 546. In some sound cards, a single DSP, which combines the DSP 525 and the DSP 545, may be provided. An output of the second DAC 546 is connected through a gain control 556 to an input of the mixer 548. An output of the mixer 548 is connected through a gain control 550 to a second power amplifier 540. An output of the power amplifier 540 is provided to the loudspeaker system 514.

The internal structure of the sound card 510 has been simplified to more effectively illustrate the use of the sound card to implement various embodiments and features of the present invention. The sound card may also have additional capabilities such as inputs connected to analog-to-digital converters (ADCs) (not shown) to allow a user to produce sampled digital data from an analog audio source. The sound card 510 may also provide input/output ports for connecting to joysticks, and MIDI input/output ports for connecting to musical instruments that have MIDI ports. The sound card 510 may also provide a line input port and a line output port, as well as input ports for audio input from devices such as CD players and Digital Audio Tape (DAT) drives. The sound card 510 may also provide DSP capabilities for programming the action of the synthesizers 524 and 544. The synthesizers 524 and 544 may be programmed by using the DSPs 525 and 544 or the sound card 510 may provided, other DSP resources for programming the action of the synthesizers 524 and 544. Some embodiments of the present invention may comprise software that runs on the DSP processors provided by the sound card 510, as shown in Figure 5.

A multimedia program loaded into the memory 504 and running on the processor 502 uses the sound card 510 to generate audio signals which are converted into sounds (acoustic energy) by the loudspeakers 512 and 514. Audio signals may be generated by sending commands to the synthesizers 524 and 544. Audio signals generated by the first synthesizer 524 are passed through the gain control stage 534, to the mixer 528, through the gain control 530, through the power amplifier 520 and subsequently turned into acoustic energy by the loudspeaker 512. A similar signal processing path, comprising the gain controls 556 and 550, the mixer 548 and the power amplifier 540 is provided for audio signals generated by the second synthesizer 544.

A multimedia program may also generate audio signals from digitized audio data by direct digital-to-analog conversion using the DACs 526 and 546. Digitized audio data may be stored on the storage media 506, or in the main memory 504. The storage media 506 may be any apparatus for storing data, including a disk drive, Compact Disk (CD), DVD, DAT drive, etc. Digitized audio data stored on the storage medium may be stored in any raw form, including Pulse Code Modulation (PCM), or any compressed form, including Adaptive Pulse Code Modulation (ADPCM). Digitized audio data stored on a hard disk or other storage medium (e.g., a CD-ROM) that provides a file system under the Microsoft Windows operating environment is generally stored in a file format known to those skilled in the art as a "wave" file having a file name *.wav (where the "*" indicates a wildcard file name).

Figure 6A is a block diagram that illustrates the process of creating sounds from a digital source 600. The digital source 600 may be any source of digitized audio including, by way of example, an analog-to-digital converter, DSP, compact disc player, laser disc players, digital versatile disk (DVD) players, devices for recording and playback of prerecorded audio, multimedia devices, computer programs, wave files, computer games and the like. Digital data is provided by the digital source 600 to a digital-to-analog converter 602 which converts the digital data into an output analog signal. The converter 602 provides the output analog signal to other analog devices such as power amplifiers, loudspeakers, other signal processors, etc.

Figure 6B is a block diagram that illustrates a sound enhancement system in accordance with one embodiment of the present invention. In the Figure 6B, data from the digital source 600 is provided to a sound enhancement block 601 which performs signal processing on the digitized sound to modify the digitized sounds to improve the perceived low-frequency response of a loudspeaker. The modified digital data from the sound enhancement block 601 is provided to the digital-to-analog conversion block 602 where the digital data is converted into analog signals. The analog signals from the block 602 are provided to other analog devices such as loudspeakers, power amplifiers, or other signal processing devices. Implementation of the signal processing in the block 601 may be provided by a general purpose digital computer, such as the processor 502, or by a DSP, such as the DSPs 525 and 545.

For example, the processing may be accomplished with software loaded into a computer's memory, with a DSP manufactured by Texas Instruments Inc. (such as the TMS320xx series), with DSPs provided by other manufacturers, with multimedia processors such as the MPACT multimedia processor supplied by Chromatic

Research Inc. or with processors such as a Pentium processor, a Pentium Pro processor, an 8051 processor, a MIPS processor, a Power PC processor, an ALPHA processor, etc.

In one embodiment, the signal processing block 602 is implemented wholly in software on the processor 502. Digital data (e.g. data from a wave file) produced by a computer program running on the processor 502 is provided to a separate signal processing program which provides the functionality represented by the block 601. The separate signal processing program modifies the digital data and provides the modified digital data to the digital-to-analog converter block 602 which may be part of the sound card 510. This pure software embodiment provides a low cost method for a user on a multimedia computer system, such as the user 202 shown in Figure 2, to extend the apparent low-frequency response of the loudspeakers attached to the multimedia computer.

In an alternative software embodiment, the processing represented by the block 601 is provided by a DSP in a sound card attached to a computer. Thus, for example, the processing represented by the signal processing block 601 may be implemented by the DSP 525 and the DSP 545 in the sound card 510 shown in Figure 5. Software embodiments of the present invention are attractive because they can be implemented at with little cost.

However, hardware embodiments are also within the scope of the present invention. Figure 7 illustrates a block diagram of a hardware embodiment of the present invention wherein the sound enhancement function is provided by a sound enhancement unit 704. The sound enhancement unit 704 receives audio signals from a signal source 702. The signal source 702 may be any signal source, including the signal source 102 shown in Figure 1, or the sound card 510 shown in Figure 5. The sound enhancement unit 704 performs signal processing to modify the received audio signals in and produces audio outputs which may be provided to loudspeakers, amplifiers, or other signal processing devices.

Signal Processing

Figure 8 is a block diagram 800 of one embodiment of the low-frequency enhancement signal processing performed by the various signal processing blocks such as the sound enhancement unit 704 shown in Figure 7, the sound enhancement block 601 shown in Figure 6B, and the sound enhancer 104 shown in Figure 1. Figure 8 may also be used as a flowchart to describe a program running on a DSP or other processor which implements the signal processing operations of an embodiment of the present invention.

Figure 8 shows two inputs, a left-channel input 802 and a right-channel input 804. The two channels of signal processing shown in Figure 8 will be conveniently described in terms of a left channel and a right channel in accordance with normal stereo left and right channels, however, the invention is not so limited and includes systems with more than two channels and systems in which the channels do not correspond to stereo left and right channels.

The inputs 802 and 804 are both provided to an adder 806 which produces an output that is a combination of the two inputs, the combination being the linear sum of the two inputs. An output of the adder 806

is provided to an amplifier 808. The gain of the amplifier 808 can be adjusted to a desired value. The adder 806 and the amplifier 808 can also be combined into a single summing amplifier that provides summing of the two inputs and gain.

An output of the amplifier 808 is provided to a lowpass filter 810. An output of the lowpass filter 810 is provided to a first bandpass filter 812, a second bandpass filter 813, a third bandpass filter 814, and a fourth bandpass filter 815. The output of each bandpass filter 812-815 is provided to an input of an amplifier 816-819 respectively, such that each bandpass filter drives one amplifier. An output of each of the amplifiers 816-819 is connected to an adder 820 which produces an output that is the sum of the outputs of the amplifiers.

The output of the amplifier 820 is provided to a first input of a left-channel adder 824 and the output of the amplifier 820 is provided to a first input of a right-channel adder 832. The left-channel input 802 is provided to a second input of the left-channel adder 824 and the right-channel input 804 is provided to a second input of the right-channel adder 832. The outputs of the left-channel adder 824, and the right-channel adder 832 are, respectively, the left and right-channel outputs of the signal processing block diagram 800.

The rolloff frequency and rate of the lowpass filter 810 are chosen to provide a suitable number of midbass harmonics above the lowest frequency that can reasonably be produced by the multimedia speakers. The bandpass filters 812-815 are chosen to shape the spectrum of the signal produced by the lowpass filter 810 in order to emphasize the harmonics of the low-frequency signals that will not be adequately reproduced by the loudspeakers. In one embodiment, the lowpass filter 810 is a second order Chebychev filter, having a rolloff of 12 dB/octave and a rolloff frequency of 200 Hz. Typically the bandpass filters will be stagger-tuned to frequencies of 100 Hz, 150 Hz, 200 Hz, and 250 Hz. In one embodiment, the bandpass filters 812-815 are second order Chebychev filters implemented as shown in Figure 9.

Figure 9 is a circuit diagram of an second order Chebychev filter having an input 902 and an output 918. The input 902 is provided to a first terminal of a resistor R1 904. A second terminal of the resistor R1 904 is provided to a first terminal of a resistor R2 906, a first terminal of an input capacitor 912, and a first terminal of a feedback capacitor 910. A second terminal of the input capacitor 912 is connected to an inverting input of an operational amplifier (op-amp) 914 and to a first terminal of a resistor R3 908. A non-inverting input of the op-amp 914 is connected to ground. An output of the op-amp 918 is connected to a second terminal of the feedback capacitor 910, a second terminal of the feedback resistor 908 and the output 918. In one embodiment, the input capacitor 912 and the feedback capacitor 910 are both 0.1 microfarad capacitors.

Table 1 lists the center frequencies and circuits values used for the bandpass filters 812-815 according to the circuit shown in Figure 9. Figure 10 illustrates the general shape of the transfer functions of the bandpass filters. Figure 10 shows the bandpass transfer functions 1002, 1004, 1006, and 1008, corresponding to the bandpass filters 812-815 respectively.

Table 1

| Filter | Frequency (Hz) | R1 (KΩ) | R2 (KΩ) | R3 (KΩ) |
|---|---|---|---|---|
| 812 | 100 | 31.6 | 4.53 | 63.4 |
| 813 | 150 | 21.0 | 3.09 | 42.46 |
| 814 | 200 | 15.8 | 2.26 | 31.6 |
| 815 | 250 | 12.7 | 1.82 | 25.5 |

The amplifiers 813, 815, 817, and 819 are set to a gain of two. Thus, the output of the mixer 820, and the signal 821, is an audio signal comprising the sum of the left and right stereo channels which have been filtered and processed in approximately the 100 Hz to 250 Hz range. This processed signal is added to the feedforward paths of the left and right stereo channels by the mixers 824 and 832 respectively. Since the signal 821 contains both left and right-channel information, adding the signal 821 back into the left and right channels will introduce some left-channel audio signal into the right channel, and vice versa. Thus, the effect is to equalize the two channels somewhat.

Figure 11 illustrates another signal processing embodiment of the sound enhancement system. The embodiment shown in Figure 11, is in many ways similar to the embodiment of Figure 8, except that in Figure 11, the four bandpass filters are driven by a monostable multivibrator 1112 which is triggered by a zero crossing detector 1110. Figure 11 shows two inputs, a left-channel input 1103 and a right-channel input 1101. As with Figure 8, the two channels of signal processing shown in Figure 11 will be described in terms of a left channel and a right channel as a convenience, but not as a limitation.

The inputs 1103 and 1101 are both provided to an adder 1102 which produces an output that is a combination of the two inputs, the combination being the linear sum of the two inputs. An output of the adder 1102 is provided to an amplifier 1103 having a gain of one. The gain of the amplifier 1103 can, however, be adjusted to any desired value. An output of the amplifier 1103 is provided to a lowpass filter 1104 having a frequency cutoff of approximately 100 Hz. An output of the lowpass filter 1104 is provided to a peak detector 1106 and an amplifier 1108 having a gain of approximately 0.05. The peak detector 1106 has a decay time constant of 0.25 milliseconds. An output of the amplifier 1108 is provided to the zero crossing detector (ZCD) 1110. An output of the ZCD 1110 is provided to a trigger input of the monostable 1112 such that the monostable 1112 is triggered each time the output of the lowpass filter 1404 passes through zero.

When triggered, the monostable 1112 produces a 150 millisecond pulse. A non-inverted output of the monostable 1112 is provided to a first input of a multiplier 1114 and to a control input of a SPST (single-pole single-throw) voltage controlled switch 1116, so that the switch 1116 is closed whenever the non-inverted output of the monostable 1112 is high. A second input of the multiplier is provided by an output of the peak detector 1106. An

output of the multiplier 1114 is provided to a first terminal of the switch 1114. A second terminal of the switch 1114 is provided to first bandpass filter 1118, a second bandpass filter 1119, a third bandpass filter 1120, and a fourth bandpass filter 1121. The output of each bandpass filter 1118-1121 is provided to an input of an amplifier 1126-1129 respectively, such that each bandpass filter drives one amplifier, each amplifier effectively having a gain of two. An output of each of the amplifiers 1126-1129 is provided to a mixer 1134 which produces an output that is the sum of the outputs of the amplifiers 1126-1129. The output of the mixer 1134 is provided to an input of a lowpass filter 1136 having a cutoff frequency of approximately 200 Hz. The highpass filters 1142 and 1144 both have a cutoff frequency of approximately 125 Hz.

An output of the mixer 1134 is provided to a first input of a left-channel adder 1140 and first input of a right-channel adder 1144. The left-channel input 1103 is provided to a second input of the left-channel adder 1140, and the right-channel input 1101 is provided to a second input of the right-channel adder 1144. The output of the left-channel adder 1140 is provided to an input of a highpass filter 1142, and an ouput of the highpass filter 1142 is provided to a left-channel output 1150. The output of the right-channel adder 1144 is provided to an input of a highpass filter 1146, and an ouput of the highpass filter 1146 is provided to a left-channel output 1144.

The system of Figure 11 generates pulses based on the zero crossings of the output of the lowpass filter 1104. The pulses are provided to the filters 1118-1121, and thereby cause the filters to "ring" producing harmonic frequencies, primarily in the 100 to 300 Hz range. Since the pulses are generated by the zero crossings of the input lowpass filtered input signal, the harmonics generated by the filters 1118-1121 are harmonics of the low-frequency components of the input waveform Thus, the system of Figure 11 generates harmonic content similar to what would be generated by the human ear if the low-frequency information was converted to acoustic energy. The generated harmonics are mixed with the normal left and right-channel information by the adders 1140 and 1144, highpass filtered to remove the remaining low-frequency signals, and then sent to the loudspeakers. The added harmonics will be interpreted by the brain of a listener as corresponding to lower-frequency content in the acoustic wave.

In yet another embodiment of the present invention, the amplifiers which are driven by the bandpass filters (e.g., the amplifiers 816-819 in Figure 8) are replaced with automatic gain control blocks that are controlled by the magnitude of the low-frequency content of the input audio signal. Before examining the signal processing elements used to accomplish said gain control, it is helpful to first examine the effect of gain control on the input and output audio signals in order to gain a better understanding of the process. This embodiment enhances the midbass harmonics (e.g., the harmonics between approximately 100 Hz and 250 Hz) in two ways. The spectrum in this region will be lifted and flattened according to the amount of energy in the input signal that is at frequencies too low for the speaker to reproduce (e.g., frequencies below 100 Hz). When there is little energy in the frequencies below 100 Hz, the spectrum will be changed very little. When there is much energy in the frequencies below 100 Hz, the spectrum will be significantly lifted and flattened in the midbass region. The lifting and

flattening is accomplished by means of an enhancement factor which is generated using automatic gain control (AGC) circuits. Note that the frequencies comprising the midbass region will vary and the frequency ranges given herein are provided by way of example and not intended to be a limitation.

Figure 12A shows how, in the presence of an input signal 1202 having a large low-frequency component, controlling of the gain of four stagger-tuned bandpass filters is used to generate an enhancement factor 1220 to accomplish this goal. The example input signal 1202, shown in the frequency domain, has a large peak near 40 Hz (e.g., the lowest note on a bass guitar). The amplitude of the spectrum of 1202 tapers down to smaller and smaller values with increasing frequency. Four bandpass curves 1204, 1206, 1208 and 1210 are used to represent the transfer functions of four bandpass filters tuned approximately to 100 Hz, 150 Hz, 200 Hz, and 250 Hz. The gain of each bandpass filter (represented by the height of each of the curves 1204, 1206, 1208 and 1210 is assumed to be controlled by a separate AGC. Each AGC is, in turn, controlled by the amplitude of the curve 1202 below 100 Hz (the sub-bass region).

In frequency ranges where the input audio spectrum has almost as much amplitude as the sub-bass region, then the AGC gain will be almost unity, as seen in the curve 1204. In frequency ranges where the input audio spectrum has much less amplitude than the sub-bass region, then the AGC gain will increase, as seen in the curve 1210. The enhancement factor 1220 is essentially the composite transfer functions represented by the curves 1204, 1206, 1208, and 1210. Figure 12C shows the effect of applying the enhanced factor 1220 to the input waveform 1202 to produce an enhanced waveform 1240. Since the waveform 1202 has a large sub-bass amplitude, the enhanced waveform 1240, as compared to the input waveform 1202, is significantly lifted and flattened in the midbass region.

Figures 12C and 12D show the same process as shown in Figures 12A and 12B, where an enhancement factor 1270 is generated from an input waveform 1252. Unlike the waveform 1202, the waveform 1252 has little low-frequency energy, and thus, the enhancement factor 1270 is smaller. An output waveform 1280 shown in Figure 12D is almost identical to the input waveform 1252 because the enhancement factor 1280 is so small.

Figure 13 is a block diagram 1300 of one embodiment of the low-frequency enhancement signal processing system which uses AGC to generate an enhancement factor. Figure 13 may also be used as a flowchart to describe a program running on a DSP or other processor which implements the signal processing operations of an embodiment of the present invention. Figure 13 shows two inputs, a left-channel input 1302 and a right-channel input 1304. As with previous embodiments, left and right are used as a convenience, not as a limitation. The inputs 1302 and 1304 are both provided to an adder 1306 which produces an output that is a combination of the two inputs.

An output of the adder 1306 is provided to an input of an amplifier 1308 having a gain of unity. An output of the amplifier 1308 is provided to a lowpass filter 1310 having a cutoff frequency of approximately 400 Hz. An output of the lowpass filter 1310 is provided to a first terminal of a potentiometer 1352, a first bandpass

filter 1312, a second bandpass filter 1313, a third bandpass filter 1314, and a fourth bandpass filter 1315. The output of each bandpass filter 1312-815 is provided to an audio signal input of an AGC 1316-1319 respectively, such that each bandpass filter drives one AGC. An output of each of the AGCs 1316-819 is connected to an adder 1320 which produces an output that is the sum of the outputs of the amplifiers.

A second terminal of the potentiometer 1352 is connected to ground and a wiper of the potentiometer is connected to a peak detector 1350. An ouput of the peak detector 1350 is provided to a control input of each of the AGCs 1316-1319.

The output of the amplifier 1320 is provided to a first input of a left-channel adder 1324 and the output of the amplifier 1320 is provided to a first input of a right-channel adder 1332. The left-channel input 1302 is provided to a second input of the left-channel adder 1324 and the right-channel input 1304 is provided to a second input of the right-channel adder 1332. The outputs of the left-channel adder 1324 and the right-channel adder 1332 are, respectively, a left-channel output 1323 and a right-channel output 1333 of the signal processing block 1300. In one embodiment, the bandpass filters 1312-1315 are substantially identical to the bandpass filters 812-815 as shown in Figure 9 and Table 1.

The AGC 1318 (as well as the AGCs 1317-1319), is essentially a linear amplifier with an internal servo feedback loop. The servo automatically adjusts the amplitude of the output signal to match the amplitude of a signal on the control input. Thus, it is the control input, not the amplifier signal input, that determines the average amplitude of the output signals. If the input signal is reduced in amplitude, then the servo will increase the forward gain of the AGC 1318 so that the output signal level remains constant.

Figure 14A is a block diagram of one embodiment of the AGCs 1318-1319, comprising an audio input 1403, a control input 1402, and an audio output 1404. The audio input 1403 is provided to an input of gain controlled amplifier 1414. An output of the amplifier 1414 is provided to the audio output 1404 and a negative peak detector 1412. An output of the negative peak detector is provided to a first input of an adder 1418 and the control input 1402 is provided to a second input of the adder 1418. An output of the adder 1418 is provided to a input of an integrator 1416, and an output of the integrator 1416 is provided to a gain control input of the amplifier 1414. Together the adder 1418 and the integrator 1416 form a summing integrator 1410.

Figure 14B is one embodiment of a circuit diagram of the AGC shown in Figure 14A. As shown in Figure 14B, the gain controlled amplifier 1414 comprises an NE572 compandor 1439 having signal pins 2-8 listed in Table 2. The audio input 1403 is provided to a first terminal of an input capacitor 1442. A second terminal of the input capacitor is connected to pin 7 of the compandor 1439. The input capacitor 1442 capacitor comprises the parallel combination of a 2.2 mf (microfarad) capacitor and a 0.01 mf capacitor. The pin 2 of the compandor 1403 is connected through a 10.0 mf capacitor 1443 to ground. The pin 4 of the compandor 1403 is connected through a 1.0 mf capacitor 1444 to ground. The pin 8 of the compandor 1439 is grounded. The pin 6 of the compandor 1439 is connected to a first terminal of a 1.0 KΩresistor 1445 A second terminal of the resistor 1445 is

connected to a 2.2 mf capacitor 1446, a non-inverting input of an op-amp 1447 and a non-inverting input of an op-amp 1452. A second terminal of the capacitor 1446 is grounded. The pin 5 of the compandor 1439 is connected to an inverting input of the op-amp 1447, a first terminal of a 17.4 KΩ feedback resistor 1449 and a first terminal of a 17.4 KΩ input resistor 1450. An output of the op-amp 1447 is connected to a second terminal of the feedback resistor 1449 and a first terminal of an output capacitor 1448. An ouput of the op-amp 1452 is connected to a second terminal of the input resistor 1450. A 10.0 KΩ feedback resistor is connected between an inverting input and the output of the op-amp 1452. A 10.0 KΩinput resistor connects the inverting input of the op-amp 1452 to ground.

The gain control input of the amplifier 1414 is provided to a first terminal of a 3.0 KΩ input resistor 1440. A second terminal of the resistor 1440 is connected to the emitter of a small-signal transistor 1441, which may be a 2N2222. The base of the transistor is connected to ground, and the collector of the transistor 1441 is connected to pin 3 of the compandor 1439.

The negative peak detector 1412 comprises an op-amp 1438 and a diode 1437. The input of the negative peak detector 1412 is connected to a non-inverting input of the op-amp 1438. An output of the op-amp 1438 is connected to the cathode of the diode 1437. The anode of the diode 1437 is connected to an inverting input of the op-amp 1437 and to the output of the peak detector 1412. The peak detector 1350, shown in Figure13 may be constructed in a manner similar to the negative peak detector 1412, except that the diode 1437 is reversed for the peak detector 1350.

The first input of the summing integrator 1410 is provided to a first terminal of the parallel combination of a 100.0 KΩ resistor 1431 and a 4.7 mf capacitor 1432. The second input of the summing integrator 1410 is provided to a first terminal of the parallel combination of a 100.0 KΩ resistor 1433 and a 4.7 mf capacitor 1434. The second terminals of both parallel combinations are connected to an inverting input of an op-amp 1435. A non-inverting input of the op-amp 1435 is grounded, and a 0.33 mf feedback capacitor 1436 is connected between the inverting input of the op-amp 1435 and the output of the op-amp 1435. The output of the op-amp 1435 is the output of the summing integrator 1410.

The NE572 is a dual-channel, high-performance gain control circuit in which either channel may be used for dynamic range compression or expansion. Each channel has a full-wave rectifier to detect the average value of input signal, a linearized, temperature-compensated variable gain cell and a dynamic time constant buffer. The buffer permits independent control of dynamic attack and recovery time with minimum external components and improved low-frequency gain control ripple distortion. Pin-outs for the NE572 are listed in Table 2 (where n,m designates channels A,B). The NE572 is used in the present embodiments as an inexpensive, low-noise, low distortion, gain controlled amplifier. One skilled in the art will recognize that other gain controlled amplifiers can be used as well.

Table 2

| Pin | Function |
|---|---|
| 1,15 | Tracking Trim |
| 2,14 | Recovery |
| 3,13 | Rectifier input |
| 4,12 | Attack |
| 5,11 | Vout |
| 6,10 | THD trim |
| 7,9 | Vin |
| 8 | Ground |
| 16 | Vcc |

Figure 15 is a diagram of a signal processing system 1500 of one embodiment of the low-frequency enhancement system which provides selectable frequency ranges. Figure 15 may also be used as a flowchart to describe a program running on a DSP or other processor which implements the signal processing operations of an embodiment of the present invention. The selectable frequency range feature embodied in the system 1500 is applicable to all of the previous embodiments. For simplicity, however, the system 1500 is shown as a modification of the signal processing system 1300 shown in Figure 13, and thus only the differences between the system 1300 and the system 1500 will be described herein. In the system 1500, the output of the bandpass filter 1315 is not connected directly to the input of the AGC 1319, as in the system 1300, but rather, the output of the bandpass filter 1315 is provided to a first throw of a single pole double throw (SPDT) switch 1562. The pole of the switch 1562 is provided to the signal input of the AGC 1319. An input of a bandpass filter 1560 is connected to the input of the bandpass filter 1315 so that the bandpass filters 1560 and 1315 receive the same input signals. An output of the bandpass filter 1560 is provided to a second throw of the SPDT switch 1562.

The bandpass filter 1560 is desirably tuned to a frequency below 100 Hz, such as 60 Hz. When the switch 1562 is on a first position, corresponding to the first throw, it selects the bandpass filter 1315 and causes the system 1500 to operate identically to the system 1300, providing bandpass filters at 100, 150, 200, and 250 Hz. When the switch 1562 is in a second position, corresponding to the second throw, it deselects the bandpass filter 1315 and selects the bandpass filter 1560, thus providing bandpass filters at, say, 60, 100, 150, and 200 Hz.

Thus, the switch 1562 desirably allows a user to select the frequency range to be enhanced. A user with a loudspeaker system that provides small woofers, such as woofer of three to four inches in diameter, will typically select the upper frequency range provided by the bandpass filters 1312-1315 which are tuned to 100, 150, 200, and 250 Hz respectively. A user with a loudspeaker system that provides somewhat larger woofers, such as

woofers of approximately five inches in diameter or larger, will typically select the lower frequency range provided by the bandpass filters 1560 and 1312-1314 which are tuned to 60, 100, 150, and 200 Hz respectively. One skilled in the art will recognize that more switches could be provided to allow selection of more bandpass filters and more frequency ranges. Selecting different bandpass filters to provide different frequency ranges is a desirable technique because the bandpass filters are inexpensive and because different bandpass filters can be selected with a single throw switch.

Other Embodiments

While certain specific embodiments of the invention have been described, these embodiments have been presented by way of example only, and are not intended to limit the scope of the present invention. For example, the present invention is not limited to embodiments where the input channels are combined to produce a combined channel which is then modified to produce enhanced bass. No combination of channels is required, and the enhancement signal processing may be performed on the separate input channels. Various embodiments used biquad and Chebychev filters, however, the invention is not limited to these filter alignments. Thus, other filter alignments may be used as well. Further, the filtering may be accomplished by using combinations of lowpass and highpass filters rather than the bandpass filters described. Accordingly, the breadth and scope of the present invention should be defined only in accordance with the following claims and their equivalents.

WHAT IS CLAIMED IS:

1. An audio system for processing left and right stereo signals containing audio information intended for reproduction by left and right loudspeakers, wherein said left and right loudspeakers cannot adequately reproduce bass in low-frequency ranges, said audio system configured to create the perception of the reproduction of enhanced bass by said left and right loudspeakers, said audio system comprising:

a left loudspeaker and a right loudspeaker, said left and right loudspeakers capable of reproducing midbass and higher frequencies more accurately than bass frequencies;

a left audio signal and a right audio signal;

a first electronic adder which combines said left and right audio signals to create a mono signal, said mono signal having a set of bass frequencies and a set of midbass frequencies;

an electronic audio enhancer in communication with said mono signal, said electronic audio enhancer configured to spectrally shape said midbass frequencies to create the illusion that said shaped midbass frequencies include said bass frequencies when said midbass frequencies are reproduced on said right and left loudspeakers;

a second electronic adder which combines said modified mono signal with said left audio signal to create a modified left output signal; and

a third electronic adder which combines said modified mono signal with said right audio signal to create a modified right output signal, wherein said modified right output signal and said modified left output signal drive said left and right loudspeakers to create the illusion that said left and right loudspeakers can adequately reproduce said bass frequencies.

2. The audio enhancement system of Claim 1, wherein said audio enhancer comprises a plurality of bandpass filters.

3. The audio enhancement system of Claim 2, wherein an output of each of said bandpass filters is provided to an automatic gain control circuit.

4. The audio enhancement system of Claim 3, wherein an output of each of said automatic gain control circuits is provided to said second electronic adder.

5. The audio enhancement system of Claim 1, wherein said electronic audio enhancer is responsive to said bass frequencies.

6. The audio enhancement system of Claim 1, wherein said electronic audio enhancer shapes said midbass frequencies in response to said bass frequencies.

7. A apparatus for enhancing audio, comprising:

a signal having a first set of frequencies and a second set of frequencies;

a signal processor configured to modify said second set of frequencies in said signal to create the perception that said second set of frequencies contains at least some of said first set of frequencies; and

a combiner to combine said modified signal with said first signal to produce an output signal.

8. A apparatus for enhancing audio, comprising:

a first combiner to combine at least a portion of a first signal with at least a portion of a second signal to create a combined signal which has a first set of frequencies and a second set of frequencies;

a signal processor configured to modify said second set of frequencies in said combined signal to create the perception that said second set of frequencies contains at least some of said first set of frequencies; and

a second combiner to combine said modified combined signal with said first signal to produce an output signal.

9. The apparatus of Claim 8, wherein said second set of frequencies are a frequencies higher than said first set of frequencies.

10. The apparatus of Claim 8, wherein said signal processor enhances frequencies in said second frequency range relative to frequencies in said first frequency range.

11. The apparatus of Claim 8, wherein said signal processor comprises a plurality of filters.

12. The apparatus of Claim 8, wherein said signal processor comprises a plurality of bandpass filters.

13. The apparatus Claim 12, wherein said bandpass filters comprise adjustable gain stages.

14. The apparatus of Claim 13, wherein said adjustable gain stages are automatic gain control amplifiers.

15. The apparatus of Claim 14, wherein said automatic gain control amplifiers are configured to provide a gain control input that is responsive to said first frequencies.

16. The apparatus of Claim 14, wherein said automatic gain control amplifiers are configured to amplify said second frequencies.

17. An apparatus for enhancing audio, comprising:

a first combiner to combine at least a portion of a first signal with at least a portion of a second signal to create a combined signal which has a first set of frequencies and a second set of frequencies;

a signal processor configured to modify said second set of frequencies in said combined signal to create the perception that said second set of frequencies contains at least some of said first set of frequencies, said signal processor comprising a plurality of bandpass filters connected in parallel; and

a second combiner to combine said modified combined signal with said first signal to produce an output signal.

18. An apparatus for enhancing audio, comprising:

a first combiner to combine at least a portion of a first signal with at least a portion of a second signal to create a combined signal which has a first set of frequencies and a second set of frequencies;

a signal processor configured to modify said second set of frequencies in said combined signal to create the perception that said second set of frequencies contains at least some of said first set of frequencies, said signal processor comprising a plurality of bandpass filters driving gain controlled amplifiers, said gain controlled amplifiers configured to produce a uniform output signal in response to an amplitude of said first set of frequencies; and

a second combiner to combine said modified combined signal with said first signal to produce an output signal.

19. An apparatus for enhancing audio, comprising:

a signal processor configured to generate a second set of frequencies from an input signal comprising a first set of frequencies, said second set of frequencies generated to create the perception that said second set of frequencies contains at least some of said first set of frequencies, said signal processor comprising a functional unit which uses an ouput of a zero crossing detector provided to an input of a monostable multivibrator to provide a series of pulses in response to said first set of frequencies, said signal processor generating said second set of frequencies by delivering said series of pulses to a plurality of bandpass filters; and

a combiner to combine said second set of frequencies with said input signal to produce an output signal.

20. An audio enhancement system comprising:

at least two audio signals, said audio signals having low-frequency spectral content in a low-frequency range which cannot be adequately reproduced by a set of loudspeakers;

a combined audio signal which is the combination of said at least two audio signals;

a spectral shaping filter configured to modify the phase and amplitude characteristics of said combined audio signal to produce a shaped signal;

a feedforward path for each of said audio signals to produce a feedforward signal; and

an adder for each of said audio signals, each adder associated with a respective feedforward path, a first input of each adder configured to receive signals from said respective feedforward path and a second input of each adder configured to receive signals from said shaped signal to produce an enhanced output signal.

21. The audio enhancement system of Claim 20, wherein said shaped signal shapes frequencies in a middle frequency range relative to frequencies above said low-frequency range.

22. The audio enhancement system of Claim 21, wherein said shaped signal de-emphasizes frequencies in said low-frequency range relative to frequencies in said middle frequency range.

23. The audio enhancement system of Claim 20, wherein said feedforward paths comprise adjustable gain stages.

24. The audio enhancement system of Claim 20, wherein said spectral shaping filter comprises a plurality of bandpass filters connected in parallel.

25. The audio enhancement system of Claim 24, wherein said bandpass filters comprise adjustable gain stages.

26. The audio enhancement system of Claim 24, wherein said bandpass filters have unequal bandwidth.

27. The audio enhancement system of Claim 24, wherein said very low-frequency range comprises frequencies below 100 hertz.

28. An audio enhancement system comprising:

an audio generator configured to output a plurality of audio signals;

a signal processing circuit coupled to said audio generator, said signal processing circuit configured to produce a combined signal and to modify selected frequency portions of said combined signal to produce a modified combined signal; and

a mixer for each of said plurality of audio signals, said mixer configured to mix said audio signal with said modified combined signal to produce a plurality of output signals.

29. The audio enhancement system of Claim 28, wherein said signal processing circuit comprises a plurality of filters.

30. The audio enhancement system of Claim 29, wherein each of said filters has a different filter transfer function.

31. The audio enhancement system of Claim 29, wherein said filters comprise bandpass filters.

32. The audio enhancement system of Claim 29, wherein said filters comprise sixth order bandpass filters, each sixth order bandpass filter configured as a series combination of three biquad filters.

33. The audio enhancement system of Claim 29, further comprising at least two loudspeakers coupled to said output signals, said loudspeakers configured to reproduce an enhanced sound image.

34. The audio enhancement system of Claim 29, wherein said filters cause said output signals to be modified such that said output filters are capable of producing a perception of enhanced low-frequency response.

35. The audio enhancement system of Claim 29, wherein said filters further comprise adjustable gain stages.

36. The audio enhancement system of Claim 35, wherein said gain stages are set to unequal gains.

37. The audio enhancement system of Claim 35, wherein said unequal gains create a modified combined signal wherein frequencies above a first range of frequencies are de-emphasized relative to frequencies outside said first range of frequencies.

38. The audio enhancement system of Claim 35, wherein said filters comprise infinite impulse response filters.

39. The audio enhancement system of Claim 35, wherein said filters comprise finite impulse response filters.

40. The audio enhancement system of Claim 35, wherein said filters are implemented in a computer processor.

41. The audio enhancement system of Claim 40, wherein said filters are specially tuned to produce a non-uniform phase and frequency response based upon second order characteristics of human perception of sound.

42. An apparatus for enhancing bass in an audio signal, comprising:

providing an audio signal comprising low frequency content;

selector means for selecting said low-frequency content of said audio signal;

filter means for filtering said low-frequency content;

amplifier means for amplifying said filtered signals; and

generating a simulated low-frequency signal by combining together said amplified filtered low frequency signals.

43. A method for enhancing bass in an audio signal, comprising the acts of:

providing an audio signal;

isolating low-frequency content of said audio signal;

filtering said low-frequency content with a plurality of bandpass filters to create a set of filtered signals;

amplifying said filtered signals; and

generating a simulated low-frequency signal by combining together said amplified filtered low frequency signals.

44. The method Claim 43, wherein each of said bandpass filters provides a unique filter transfer function.

45. The method Claim 43, wherein said act of filtering further comprises amplifying said low-frequency content.

46. The method of Claim 45, wherein said act amplifying said low-frequency content comprises amplifying an output of one or more of said bandpass filters.

47. The method of Claim 45, wherein said act amplifying said low-frequency content comprises amplifying an output of one or more of said bandpass filters using an automatic gain controlling amplifier, wherein a output level of said automatic gain controlling amplifier is responsive to one or more selected frequency components of said audio signal.

48. The method of Claim 47, wherein said selected frequency components comprise frequencies lower than said low-frequency content.

49. A method for enhancing audio, comprising the acts of:

receiving a signal having a first set of frequencies and a second set of frequencies;

modifying said second set of frequencies to create the illusion that said second set of frequencies contains at least some of said first set of frequencies; and

combining said modified second set of frequencies with said signal to produce an output signal.

50. A method for enhancing audio comprising the acts of:

combining at least a portion of a first signal with at least a portion of a second signal to create a combined signal which has a first set of frequencies and a second set of frequencies;

modifying said second set of frequencies in said combined signal to create the perception that said second set of frequencies contains at least some of said first set of frequencies; and

combining said modified combined signal with said first signal to produce an output signal.

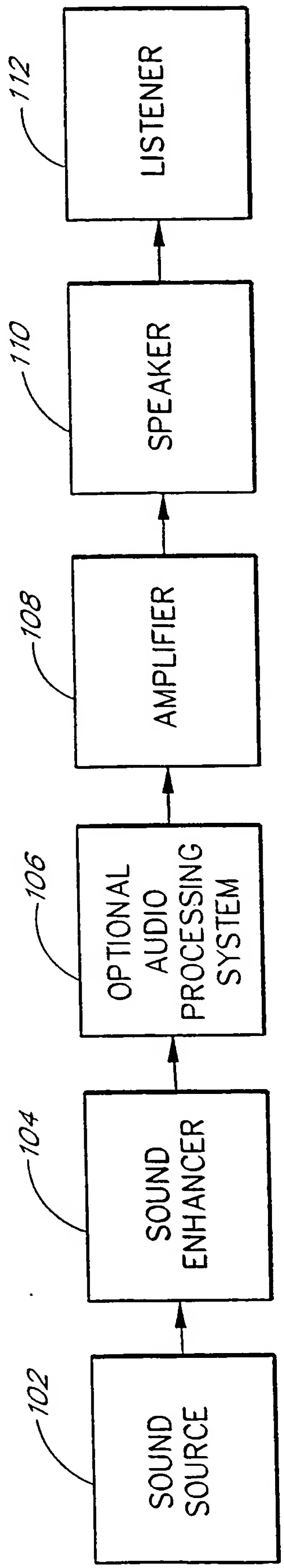

SOUND SOURCE
102
SOUND ENHANCER
104
OPTIONAL AUDIO PROCESSING SYSTEM
106
AMPLIFIER
108
SPEAKER
110
LISTENER
112

FIG. 1

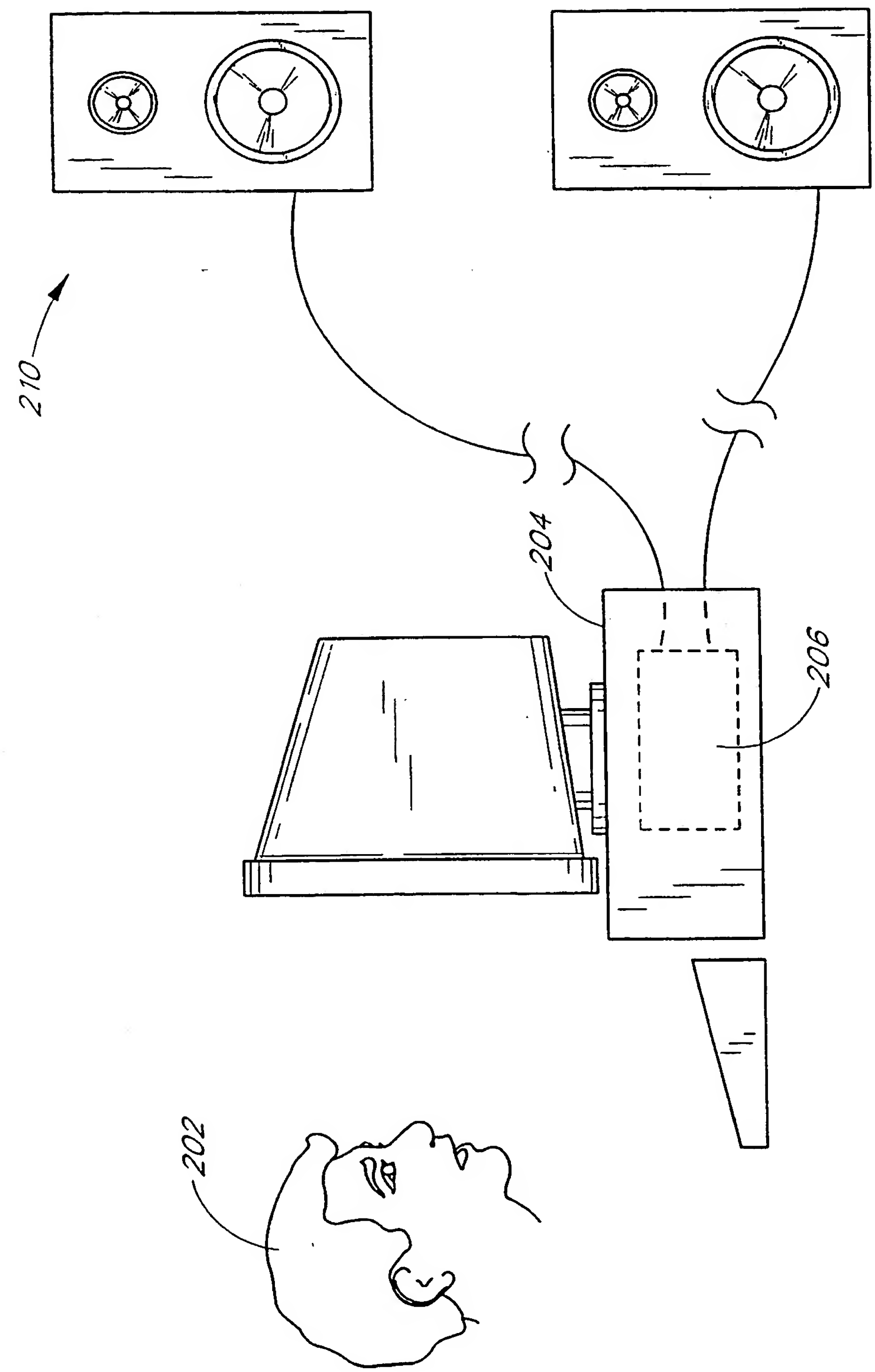
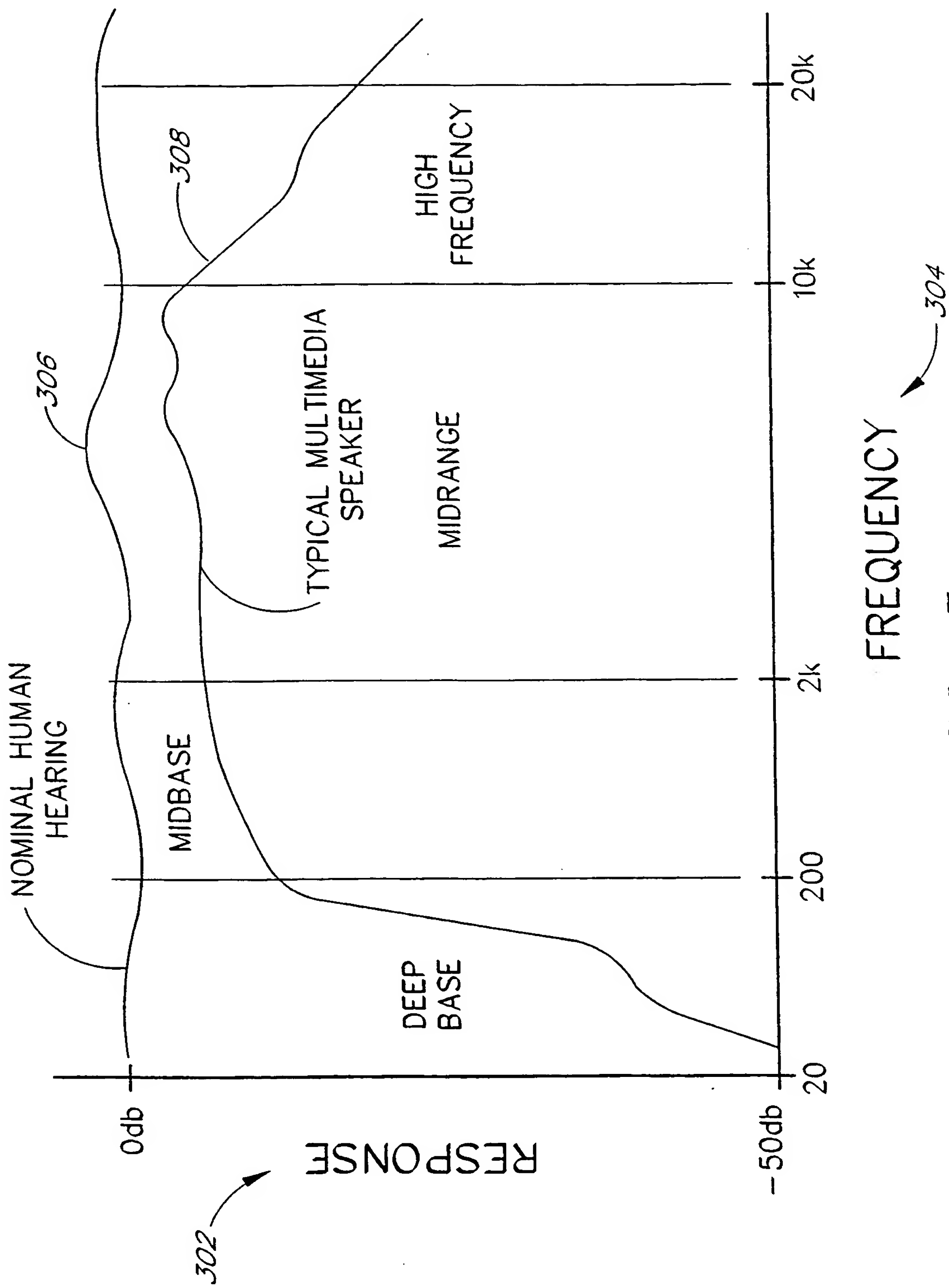



FIG. 2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TYPICAL MULTIMEDIA SPEAKER
DEEP BASE
MIDBASE
MIDRANGE
HIGH FREQUENCY
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0db
-50db
20
200
2k
10k
20k
RESPONSE
302
FREQUENCY
304

FIG. 3

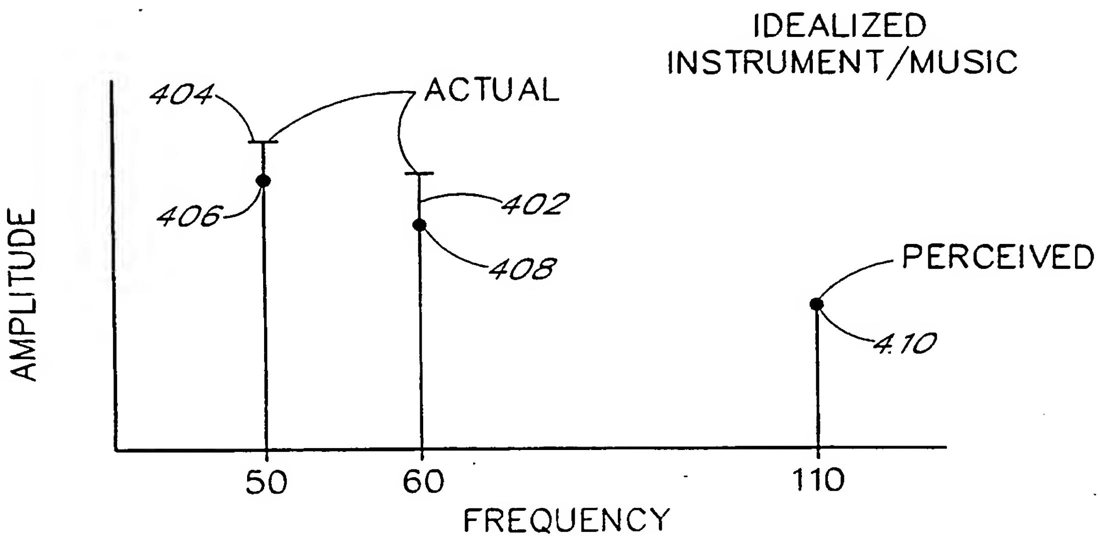
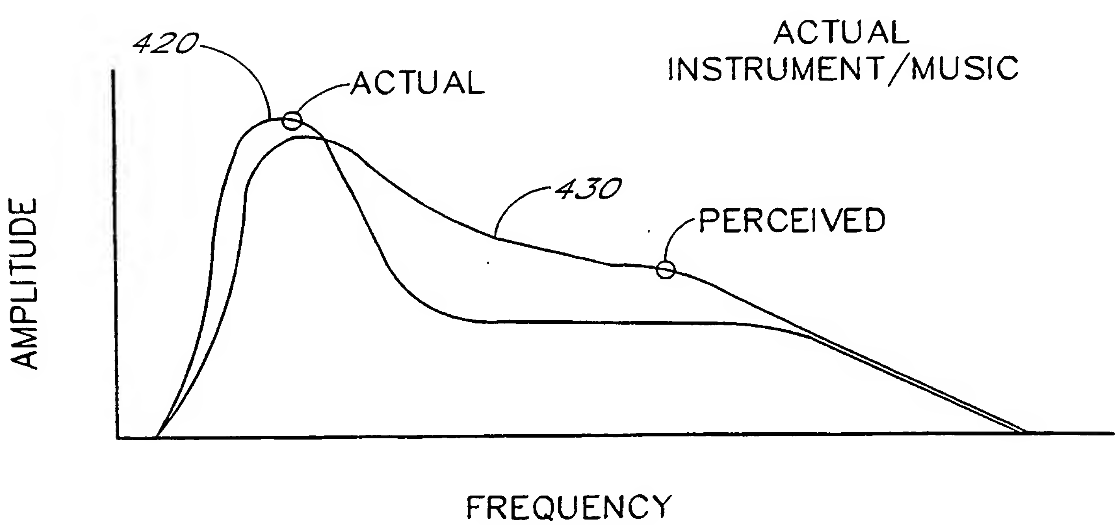

IDEALIZED
INSTRUMENT/MUSIC
ACTUAL
404
406
402
408
PERCEIVED
410
AMPLITUDE
50
60
110
FREQUENCY

FIG. 4A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INSTRUMENT/MUSIC
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ACTUAL
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AMPLITUDE
FREQUENCY

FIG. 4B

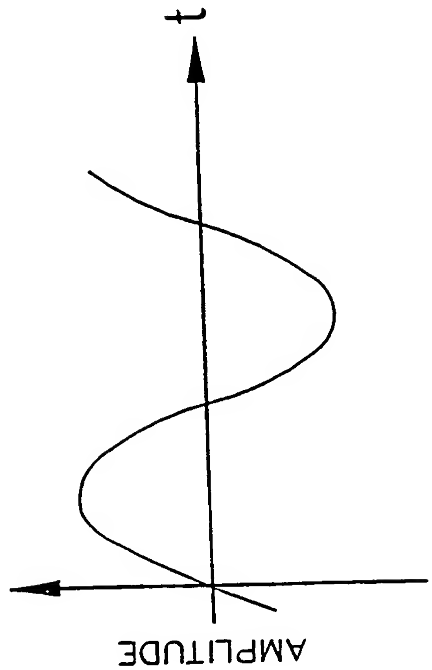



FIG. 4D

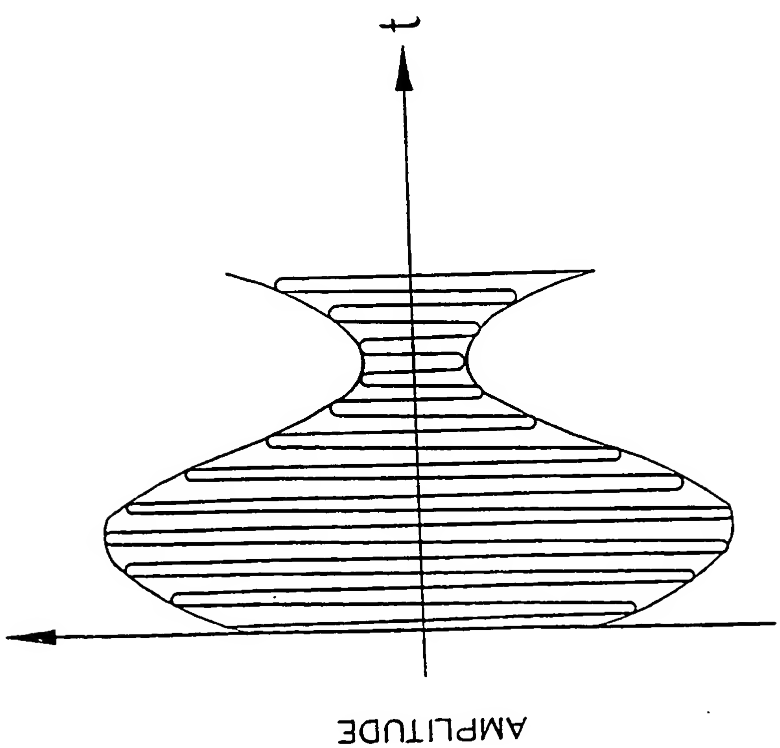
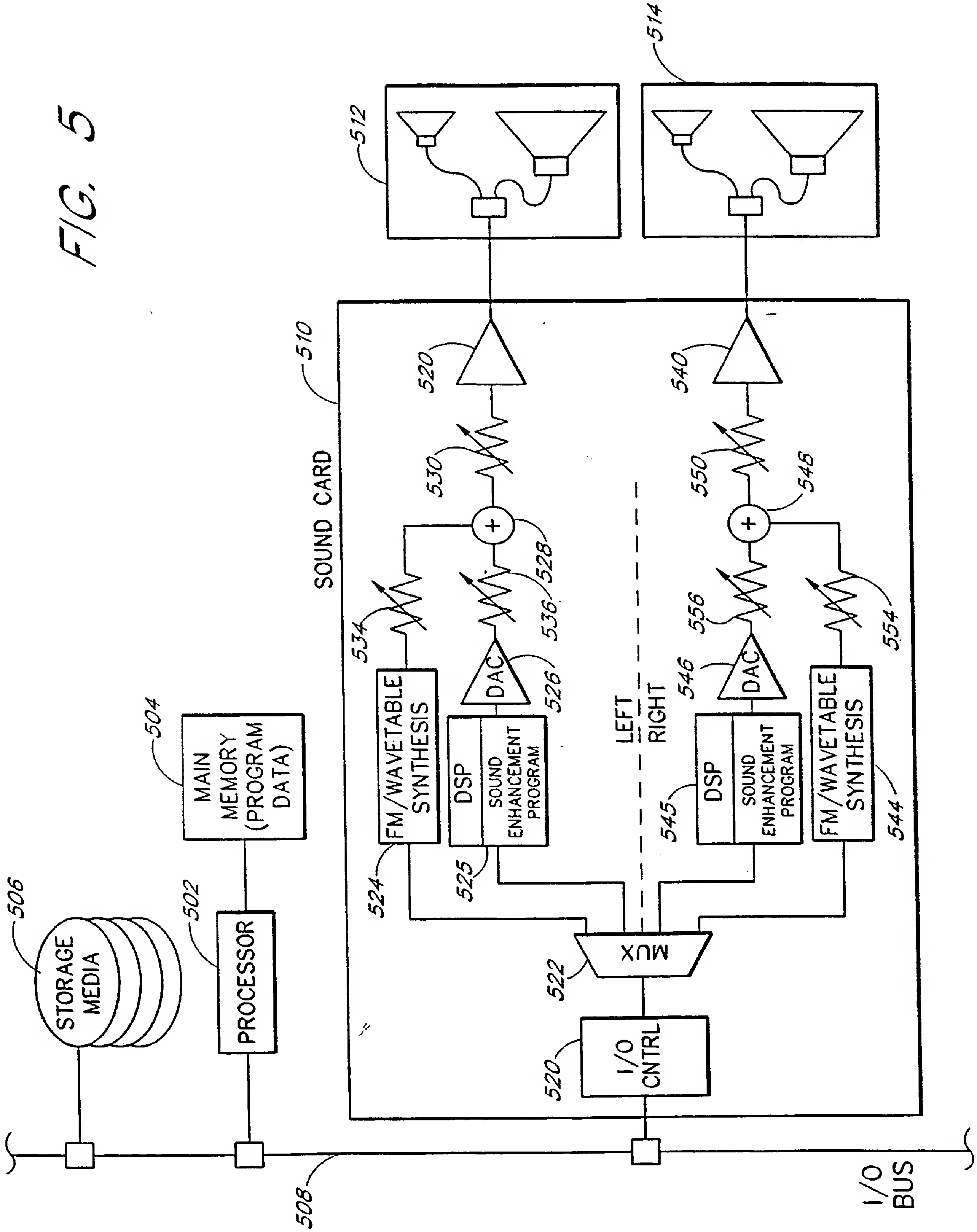



FIG. 4C

FIG. 5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514
510
SOUND CARD
520
540
530
550
548
528
536
534
556
554
526
546
DAC
DAC
LEFT
RIGHT
504
MAIN MEMORY (PROGRAM DATA)
FM/WAVETABLE SYNTHESIS
DSP
SOUND ENHANCEMENT PROGRAM
DSP
SOUND ENHANCEMENT PROGRAM
FM/WAVETABLE SYNTHESIS
545
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STORAGE MEDIA
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PROCESSOR
524
525
MUX
522
I/O CNTRL
520
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I/O BUS

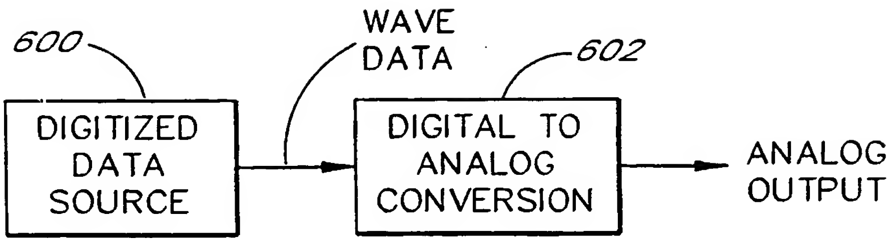



FIG. 6A
(PRIOR ART)

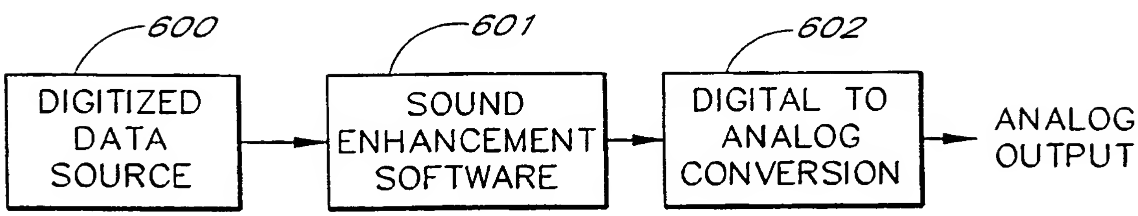
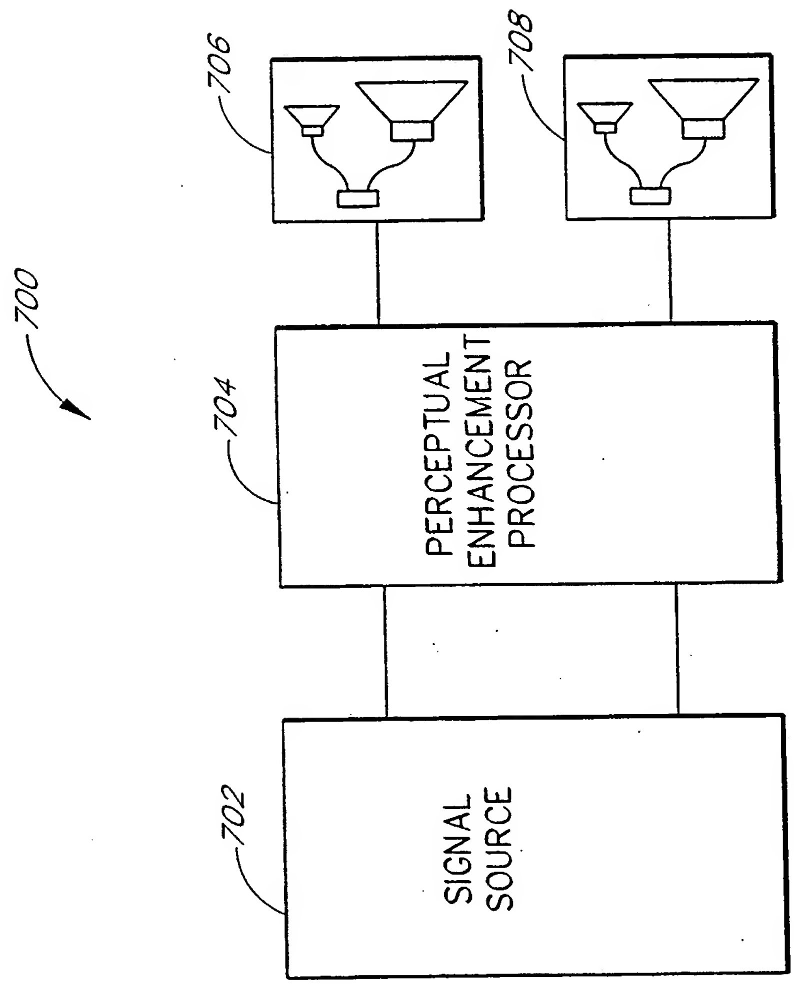



FIG. 6B


700
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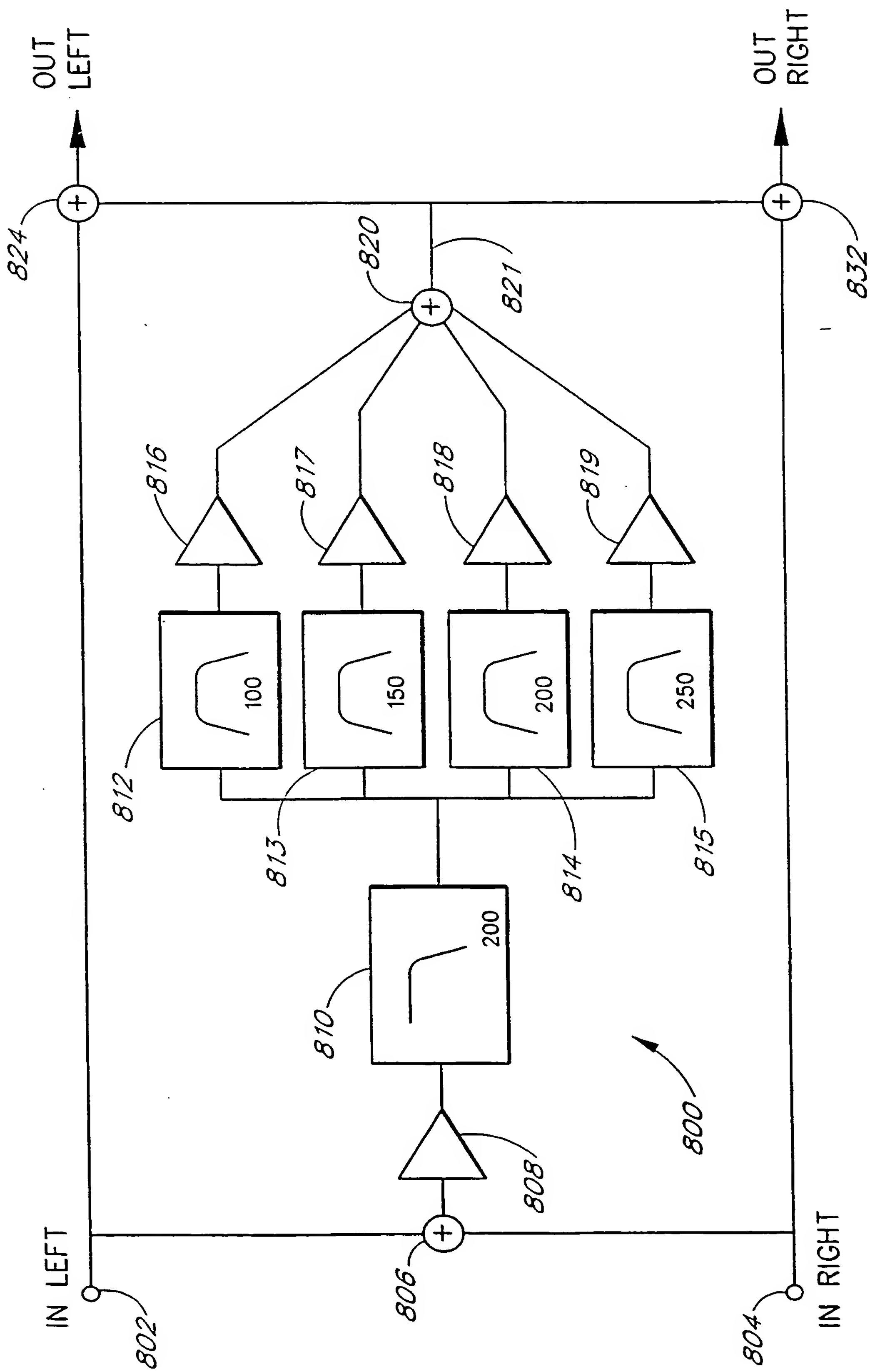
708
704
PERCEPTUAL
ENHANCEMENT
PROCESSOR
702
SIGNAL
SOURCE

FIG. 7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OUT RIGHT
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820
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832
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817
818
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806
802
804
IN LEFT
IN RIGHT

FIG. 8

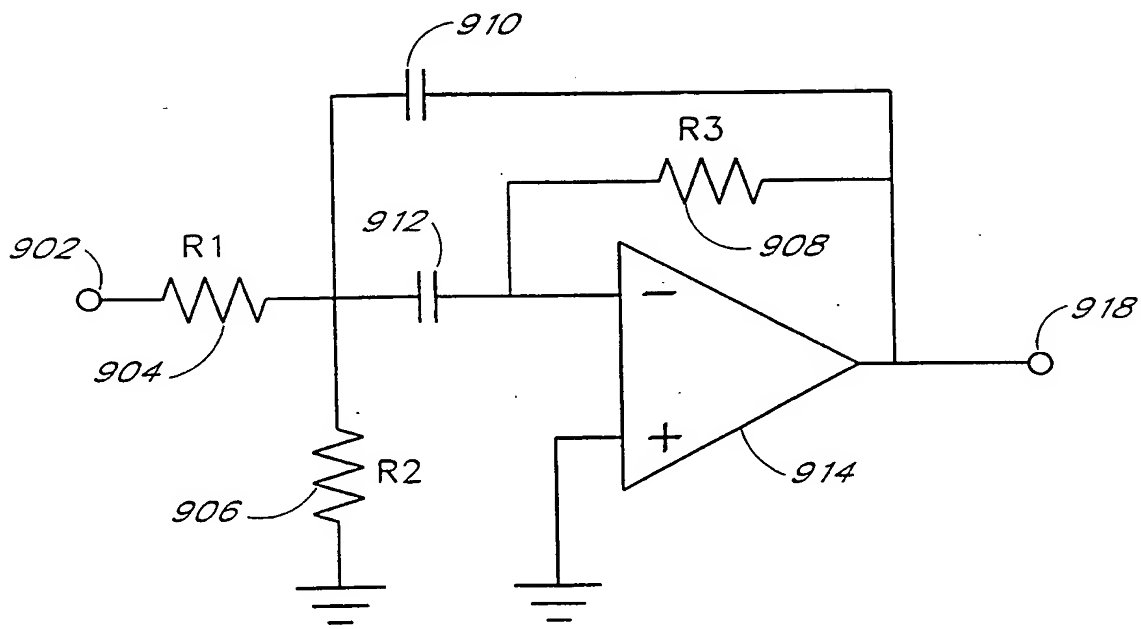



FIG. 9

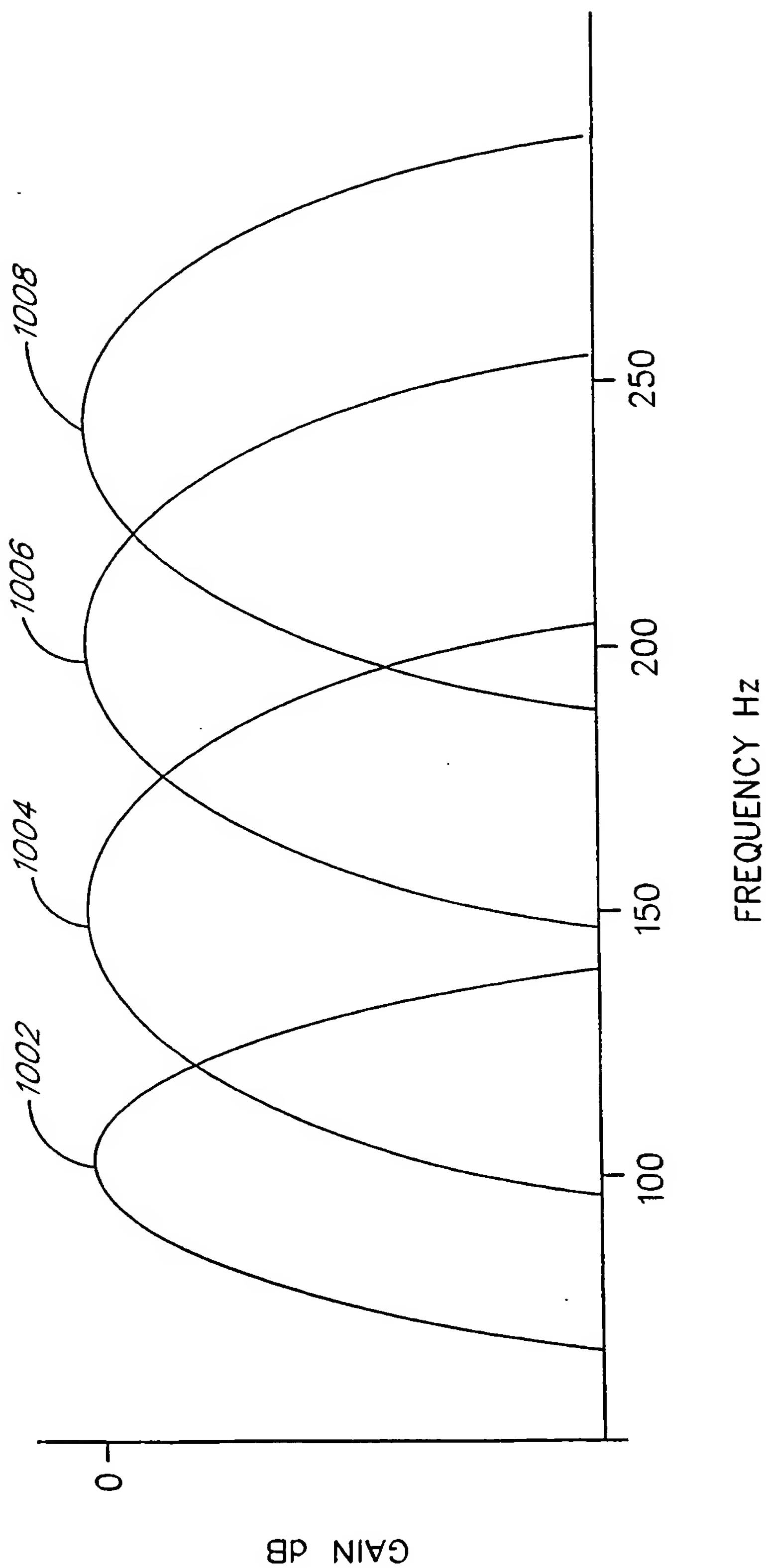
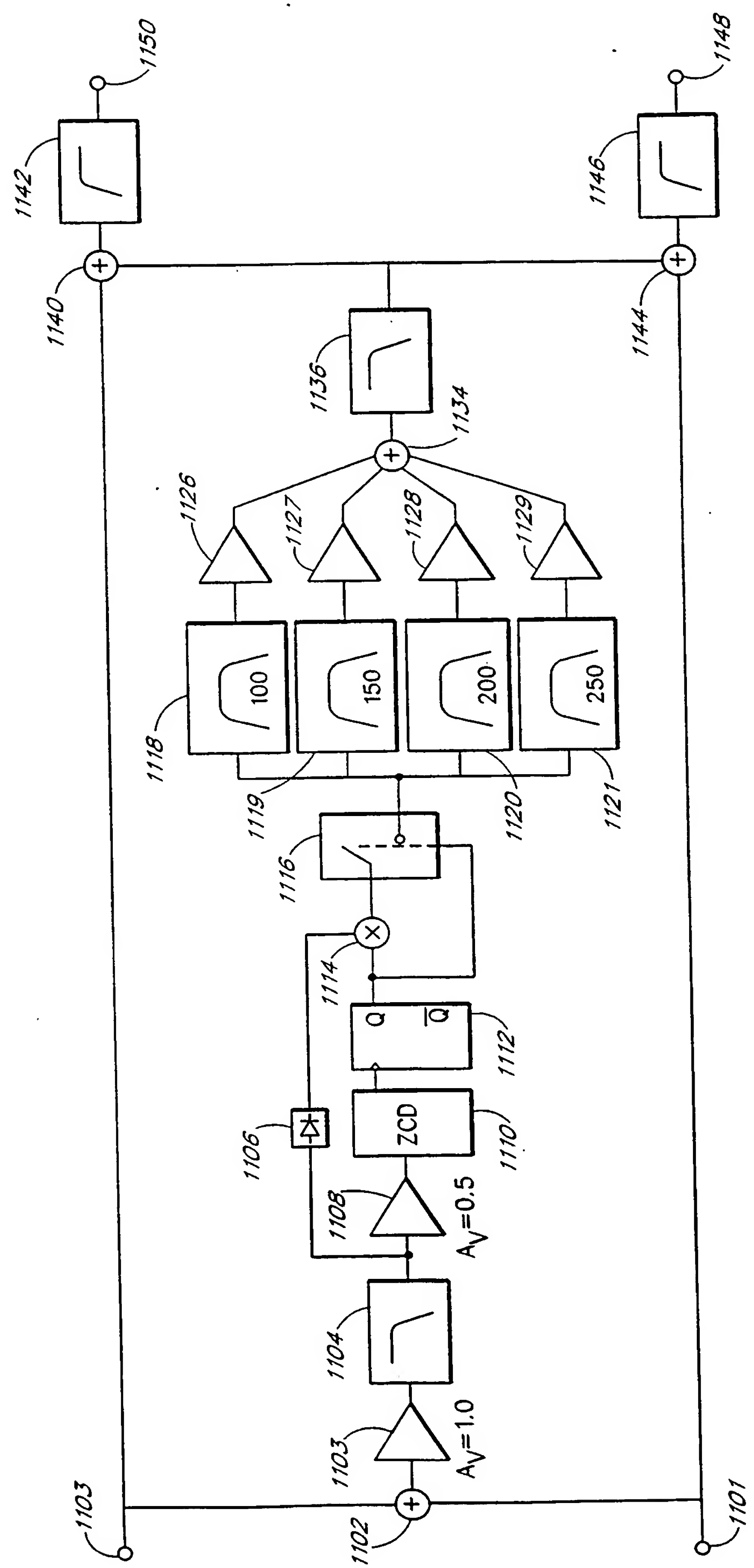



FIG. 10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1129
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200
250
1118
1119
1120
1121
1116
1114
Q
$\overline{Q}$
1112
ZCD
1110
1106
1108
$A_V$=0.5
1104
1103
$A_V$=1.0
1103
1102
1101

FIG. 11

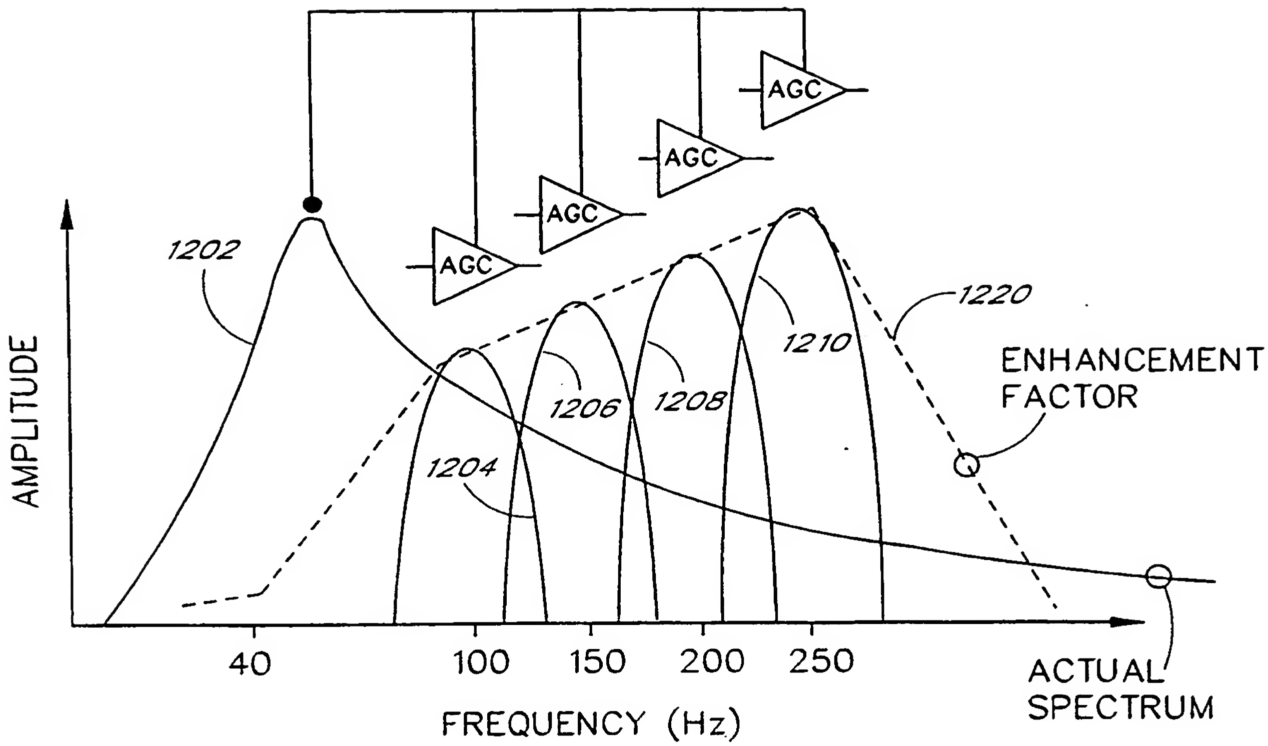


FIG. 12A

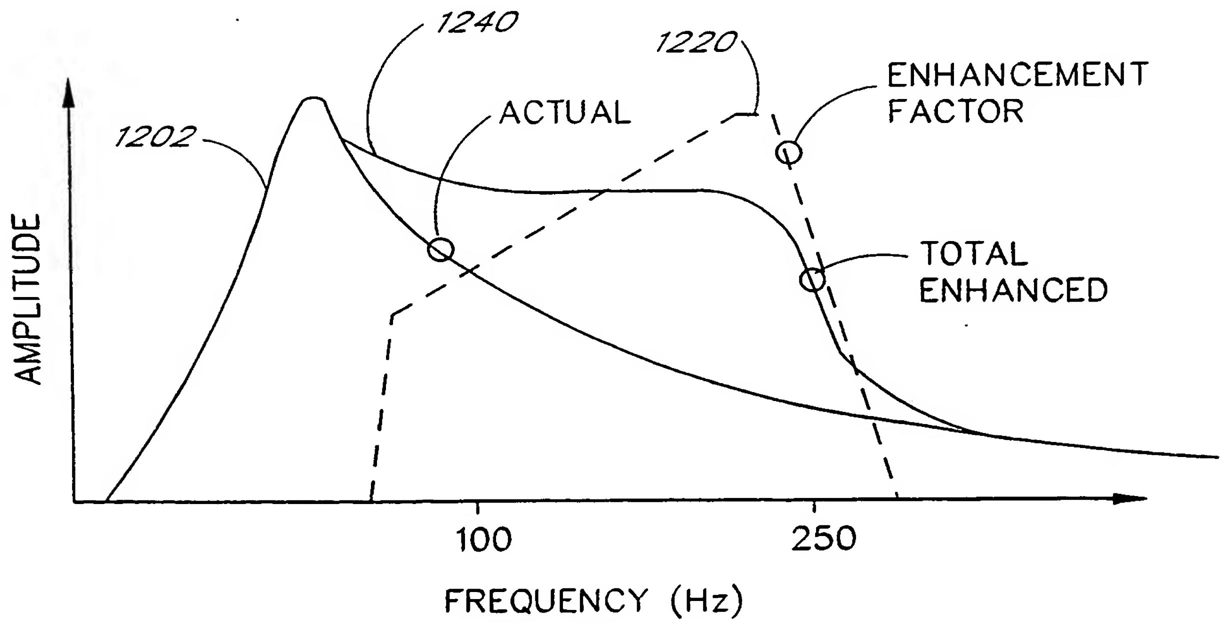


FIG. 12B

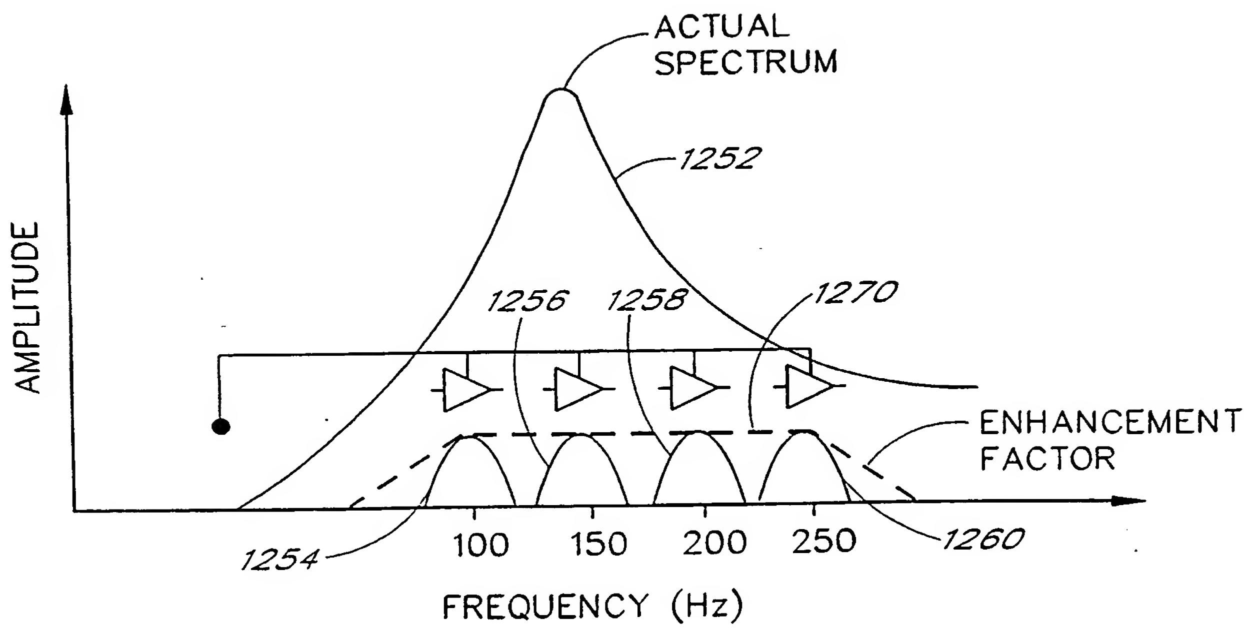


FIG. 12C

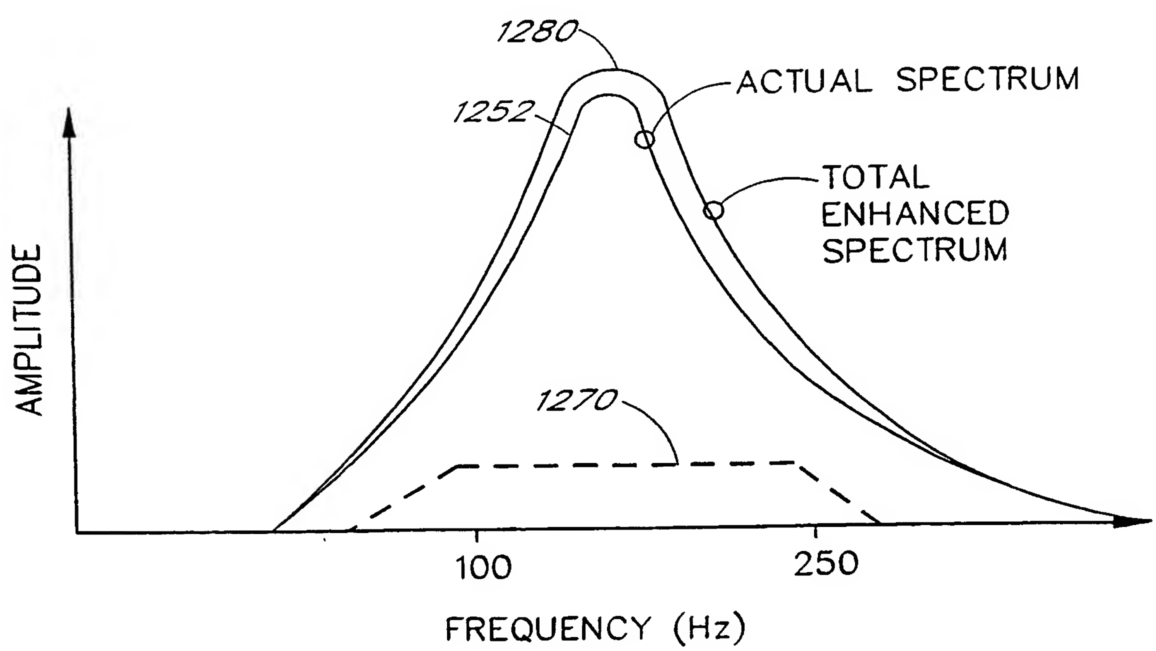


FIG. 12D

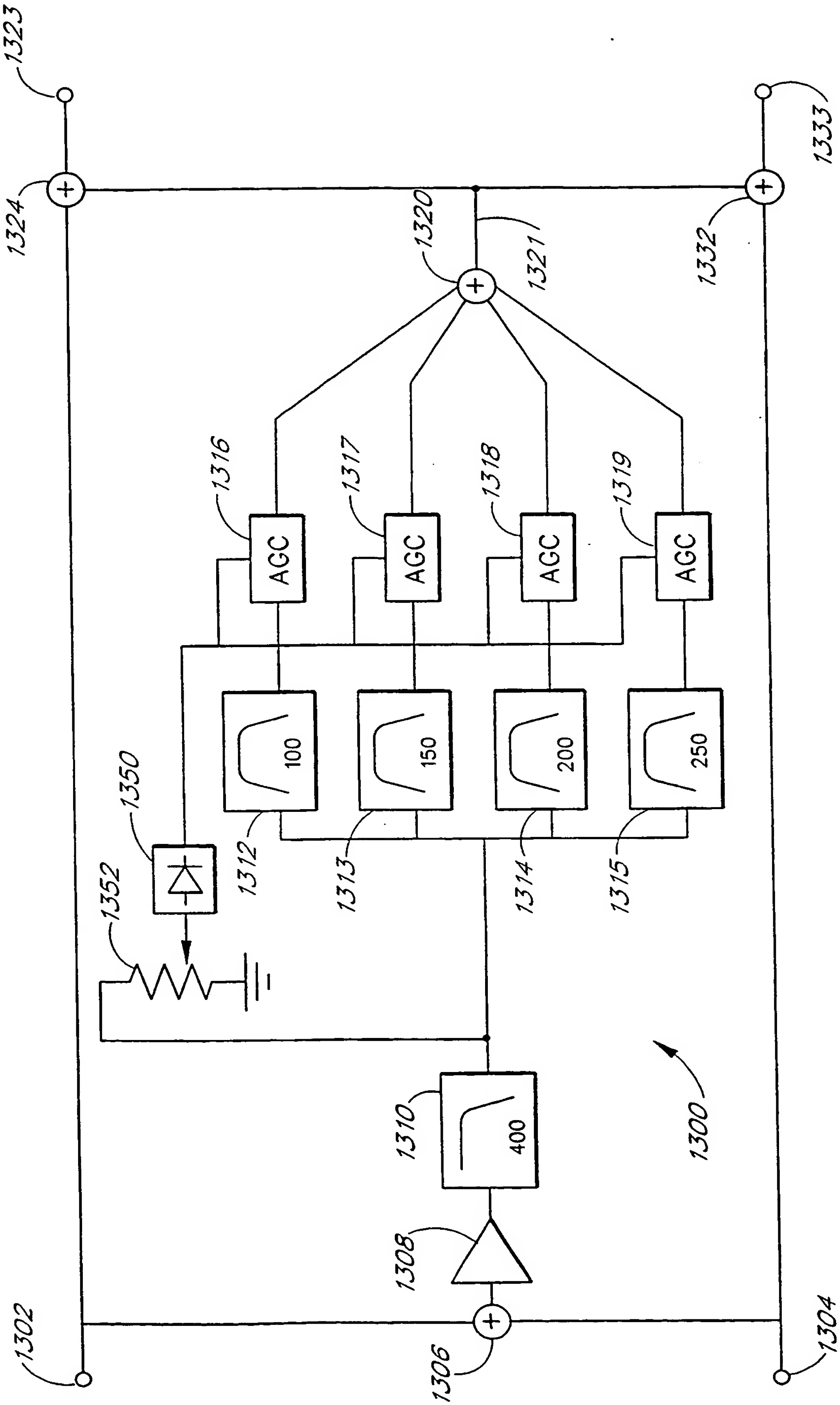
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FIG. 13

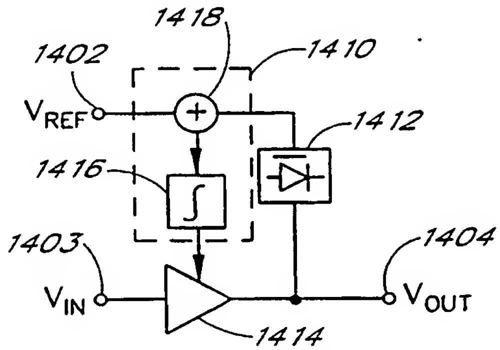


FIG. 14A

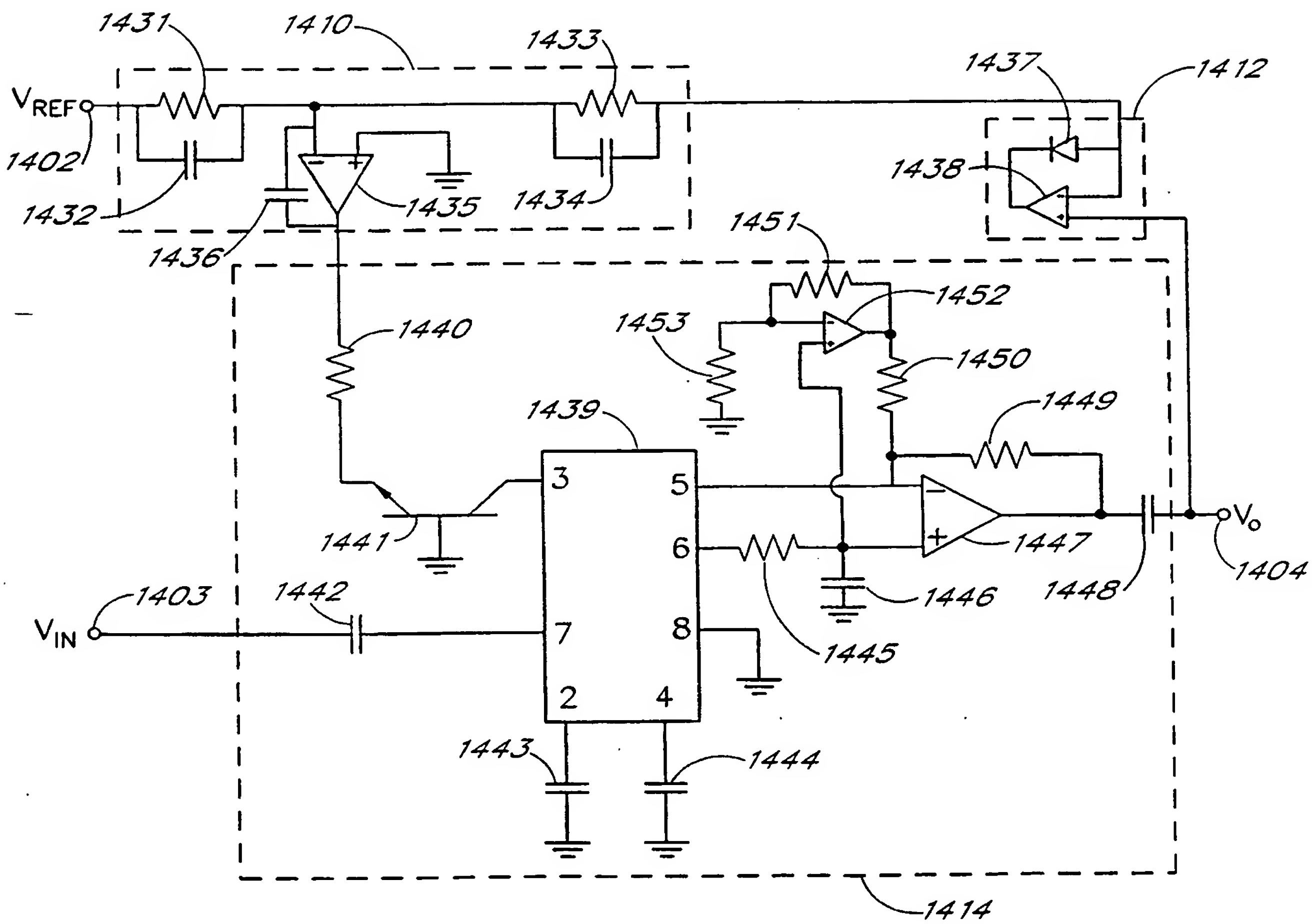


FIG. 14B

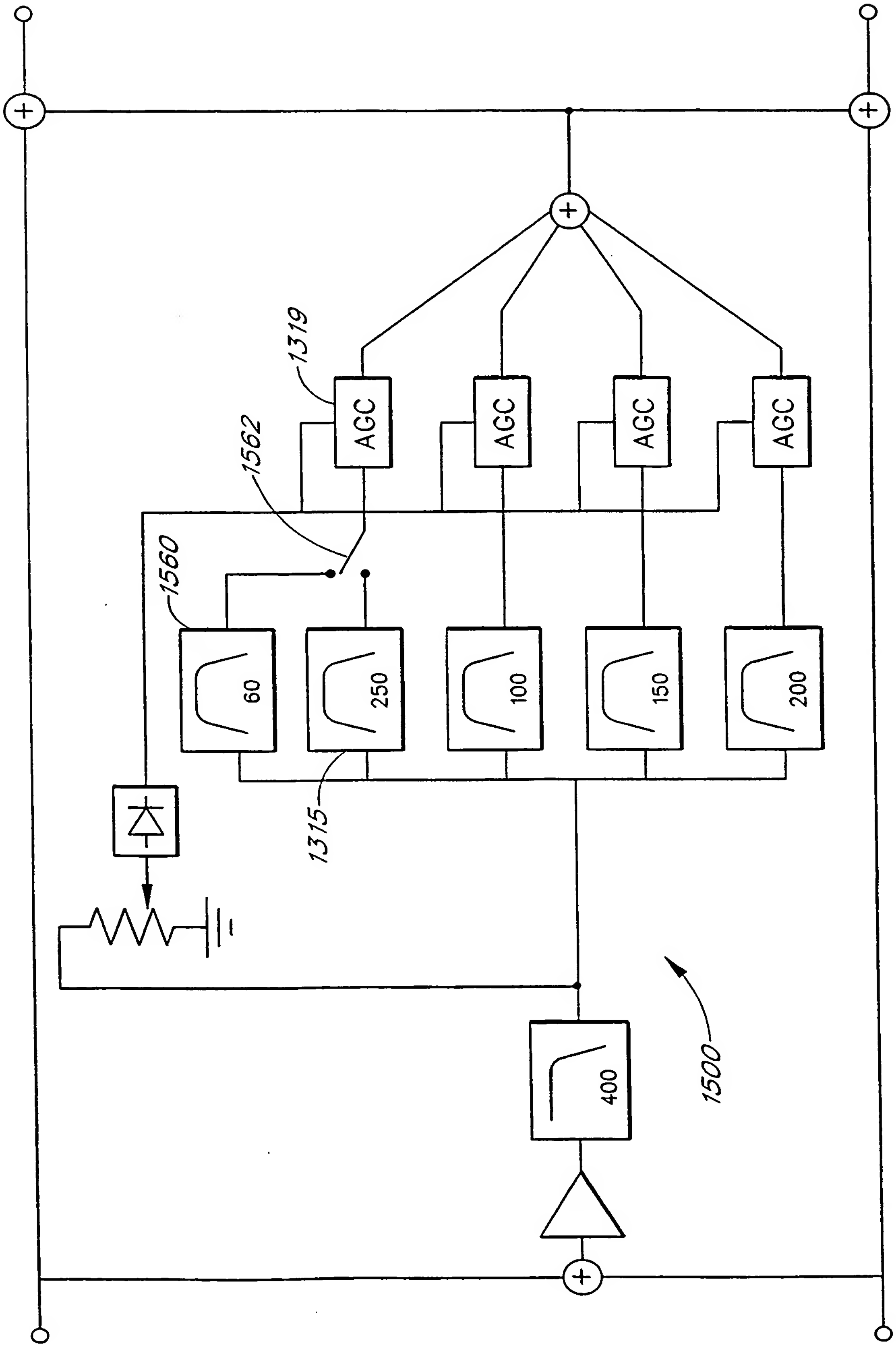


FIG. 15

# INTERNATIONAL SEARCH REPORT

International Application No
PCT/US 98/21208

**A. CLASSIFICATION OF SUBJECT MATTER**
IPC 6 H04S1/00 H04R3/04

According to International Patent Classification (IPC) or to both national classification and IPC

**B. FIELDS SEARCHED**

Minimum documentation searched (classification system followed by classification symbols)
IPC 6 H04S H04R H03G

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)

**C. DOCUMENTS CONSIDERED TO BE RELEVANT**

| Category ° | Citation of document, with indication, where appropriate, of the relevant passages | Relevant to claim No. |
|---|---|---|
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| X | EP 0 546 619 A (KONINKL PHILIPS ELECTRONICS NV) 16 June 1993<br>see page 2, line 34 - line 48<br>see page 3, line 1 - page 4, line 29; figures<br>--- | 1,28,42, 43,49,50 |
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[X] Further documents are listed in the continuation of box C.

[X] Patent family members are listed in annex.

° Special categories of cited documents :

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"E" earlier document but published on or after the international filing date

"L" document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)

"O" document referring to an oral disclosure, use, exhibition or other means

"P" document published prior to the international filing date but later than the priority date claimed

"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention

"X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone

"Y" document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art

"&" document member of the same patent family

| Date of the actual completion of the international search | Date of mailing of the international search report |
|---|---|
| 11 March 1999 | 18/03/1999 |

| Name and mailing address of the ISA | Authorized officer |
|---|---|
| European Patent Office, P.B. 5818 Patentlaan 2<br>NL - 2280 HV Rijswijk<br>Tel. (+31-70) 340-2040, Tx. 31 651 epo nl,<br>Fax: (+31-70) 340-3016 | Gastaldi, G |

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Form PCT/ISA/210 (second sheet) (July 1992)

C.(Continuation) DOCUMENTS CONSIDERED TO BE RELEVANT

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| X | see column 2, line 26 - column 4, line 17 | 28 |
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| A | US 5 359 665 A (WERRBACH DONN R) 25 October 1994 see column 1, line 50 - column 2, line 2; figures | 1,20 |
| P,X | US 5 771 296 A (UNEMURA TOYOAKI) 23 June 1998 see column 2, line 39 - column 3, line 56 see column 4, line 11 - column 5, line 40; figures | 1,28 |
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5

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Information on patent family members

International Application No
PCT/US 98/21208

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